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Preface

The Proceedings at hand include the papers presented at the 2\textsuperscript{nd} International Conference and Exhibition on Underwater Acoustics, UA2014, which was born out of the merger of the ECUA and UAM Conferences.

It is the ambition of the Scientific Committee and the Organizers of UA2014 to be able to continue the long history of successes of the ECUA and UAM Conferences, which have always attracted a large number of scientists and engineers from Europe, the United States, Canada, China, Japan etc, and have become an established forum for the presentation of the latest developments in all major areas of Underwater Acoustics.

The success of the conference is due to the joint efforts of many people. We would like to thank the Structured Session Organizers and the Scientific Committee for their valuable contribution. Our gratefulness also goes to the Office of Naval Research Global, Teledyne-RESON, and IACM-FORTH for their continuous support and Evologics GmbH for their financial contribution to this year’s event. We also appreciate the sponsoring of UA2014 by two major societies, EAA and ASA.

We are always very happy for our cooperation with the UK Institute of Acoustics and for hosting their medal awards.

Last but not least, we are grateful to all the speakers for their great and important work prepared for and presented in this conference.

We trust that these proceedings represent the state-of-the-art of most areas of Underwater Acoustics and will be a valuable reference to future works in this field.

John S. Papadakis and Leif Bjørnø
Conference Chairmen
Conference Chairmen

John S. Papadakis and Leif Bjørnø

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Session 1

Acoustic imaging

Organizer: Jiyuan Liu
A DESIGN PHILOSOPHY OF PORTABLE, HIGH-FREQUENCY IMAGE SONAR SYSTEM

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Abstract- A portable, high-frequency sonar system for 2D imaging is developing in Institute of Acoustics, Chinese Academy of Sciences. It consists of bistatic transducer arrays (separate transducer and 80-element receive hydrophone array), signal acquisition and processing unit, weak signal amplifier unit, power amplifier unit and so on. All the units are packaged in a watertight vessel. Through Ethernet port, the portable sonar system can be integrated into low power, compact marine vehicles. Prototype has been built and tested in water. It's main acoustic feature: Frequency: 550 kHz; number of Beams: 80; beam width: 45° × 15°; beam spacing: 0.75°; range resolution: 40mm. Pulse-echo with a high signal-to-noise ratio (SNR) of more than 10dB has been achieved. Due to high resolution and possessing good imaging capabilities, this sonar system has dual-use application in both military and commercial markets. The paper presents the experimental facilities, some of which are still under development, as well as the results of trials and scope of future work.

Key words: high frequency, imaging sonar; portable.
I. INTRODUCTION

A portable, high-frequency sonar system for 2D imaging is developing in Institute of Acoustics, Chinese Academy of Sciences. The system goal for 2D sonar system is to provide a diver or autonomous-underwater vehicle (AUV) with the highest possible imaging capability while minimizing space and power requirements.

The portable sonar system, possessing good imaging capabilities has dual-use application in both military and commercial markets. The potential commercial applications include channel mapping for dredging, vehicle navigation for vessels traversing relatively shallow waters, aiding search and rescue dive teams in almost every country[1].

In market similar products are available such as V Series Sonar from BlueView corporation. The V Series sonar family offers 2 field-of-view options for high-performance forward looking imaging sonar: 90°, and 130°[2]. Compare to the products from BlueView corporation , our products will be cheaper, smaller volume. Especially through Standard Ethernet Interface, it can provide client with primitive data for subsequent processing.

This paper is organized as follows. In Section II, the system is described, the design and implement of bistatic transducer arrays and electronics system are provided in Section III and the experimental facilities, some of which are still under development, as well as the results of laboratory trials and scope of future work is provided in Section IV and the future work is given in Section V.

II. SYSTEM DESCRIPTION

The portable sonar system consists of bistatic transducer arrays (separate transducer and 80-element receive hydrophone array), electronics system (Signal Conditioning Module, power amplifier Module and FPGA-based Data Acquisition and DSP Module), power, mechanical structure and so on. All modules are packaged in a watertight vessel. Through Ethernet port, the portable sonar system can be integrated into low power, compact marine vehicles. For example it can easily be mounted on the underside of the AUV. This configuration provides a downward-looking sonar with high-resolution terrain mapping. The block diagram of the portable sonar system is show in Fig. 1.
III. DESIGN AND IMPLEMENT

(1). bistatic transducer arrays

Arrays have been designed to comply with high frequency imaging applications. bistatic transducer arrays have two parts: transducer and 80-element receive hydrophone array. The schematic diagram of bistatic transducer arrays is show in the left of Fig.3. The picture of bistatic transducer arrays is show in the right of Fig.3.

Fig.3. The block diagram of the portable sonar system

Fig.3. the schematic diagram and picture of bistatic transducer arrays

In the left of Fig.3 the arc-shaped part is transducer with 10-element uniformly distributing on the arc. Due to the special arc shape, the transducer achieve a horizontal/vertical beamwidth of 45°/15°. The pulse used in the following experiments was a chirp swept from 550kHz to 600kHz. The primary source level is 190 dB. In the left of Fig.3 the shadow part is 80-element receive hydrophone array.
Due to the linear array, it provides the basis for beamforming. The hydrophonic sensitivity is -195 dB. The receiver modules is mounted 1 cm beneath the transducer.

(2). ELECTRONICS SYSTEM

The main tasks of electronic system are control, data acquisition, beamforming and communication. There are mainly three modules in this electronic system: FPGA-based multi-channel data acquisition and DSP module, signal conditioning module and power amplifier module. FPGA-based multi-channel data acquisition and DSP module is the core of the electronic system. FPGA is responsible for mission scheduling and communication. FPGA also controls the data acquisition, decimating and beamforming.

2.1 Signal Conditioning Module

Prior to beam forming, each hydrophone’s output is amplified and band-pass filtered using a low-noise amplifier with a fixed gain 20 dB. An instrumentation amplifier (LT6233) is utilized to design butterworth active filter and amplify the output voltage from a hydrophone. The LT6233 is single low noise, rail-to-rail output unity gain stable op amps that feature 1.9nV/Hz noise voltage and a 60MHz gain bandwidth product[3]. Figure 4 is a photograph of the Signal Conditioning printed circuit board.

![Figure 4 a photograph of the Signal Conditioning printed circuit board.](image)

2.2 Power amplifier module

To promote source level to 190 dB, power amplifier is indispensable. Due to high power bandwidth-2MHz[4], PA107DP is choosed as the core chip of power amplifier module. Figure 5 shows the schematic diagram of power amplifier. Rf is tunable resistor to modify gain.
2.3 FPGA-based Data Acquisition and DSP Module

In portable sonar system, FPGA-based Data Acquisition and DSP module plays a very important role in scheduling missions, beamforming and communication. Scheduling missions include a multi-channel data acquisition and the control of the power amplifier.

Due to high integration, powerful computing ability, design flexibility and user-programmability, FPGA-Xilinx Virtex-5 C5VLX220T is utilized as core controller. Xilinx Virtex-5 device is a high performance FPGA particularly suited to DSP applications[5]. Coping with high data rate, 6 FPGAs are utilized, 5 FPGAs operate synchronously for multi-channel data acquisition and beamformer, 1 FPGA operate for missions scheduling and communication.

2.3.1 multi-channel data acquisition unit

For beamforming, a multi-channel data acquisition is indispensable. The output voltage from a hydrophone will be sampled by a 16-bit 10 Msps ADC. The total data rate is about 10*80*16 Mbit/sec. According to the principle of Nyquist, all acquired data no need to be stored in SDRAM. Considering the transducer's upper working frequency (600kHz), FPGA will save 1 point every 8 points which means the true sampling rate is about 1.25M/sps. Then, the decimated data are continuously recorded in the SDRAM for subsequent processing.

2.3.2 Beamforming

A delay-and-sum beamformer allows a transducer array to "look" for signals propagating from a particular direction. By adjusting the delays associated with each element of the array, the array's look direction can be electronically steered toward the source of radiation.

Utilizing the on-chip resource, a 16-channels beamformer is implemented on a single Virtex-5 FPGA. 5 FPGAs operate synchronously for 80-channels beamformer. Via LVDS high speed serial interface, the 5 FPGAs synchronously communicate with a FPGA which is responsible for missions scheduling and communication. The block diagram of FPGA-based DSP unit is represented in Figs 7.

Figure 6 The schematic diagram of power amplifier
2.3.3. missions scheduling and communication

Making use of the integrated resource on FPGA chip, FPGA can periodically generate control signals to open power amplifier and start multi-channel data acquisition. The data are then decimated to produce 80 channels Beamforming data. Utilizing IP core, a 100M Ethernet interface is generated and FPGA communicate with an host computer. Supported by a GPU, image are displayed on the screen and beamforming data are stored in the host computer.

IV. LABORATORY EXPERIMENT RESULTS

In the spring of 2014, laboratory experiment was conducted in a pool. The configuration of the laboratory experiment is shown in Fig.8. In Fig.8 the transducer array is deployed on the right of the pool, the hydrophone array is deployed on the left of the pool, all in the depth of 20cm. The distance between source and receiver is about 0.5m. Fig.9 gives the received signal waveforms of all 32 channels. Fig.10 gives the Lisajous curve.
Figure 8. The configuration of the laboratory experiment

Figure 9. The received signal waveforms of all 32 channels
V. FUTURE WORK

Now the FPGA-based Data Acquisition and DSP Module is under joint debugging. Future work includes system debugging, lake trial and sea trial. Furthermore, we will explore the merits of different transmit waveforms and their impact on image quality.

REFERENCES

SYNTHETIC APERTURE SONAR IMAGES MOSAIC BASED ON SIFT AND RANSAC METHOD

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Abstract: Combing a number of acoustic pings, the Synthetic Aperture Sonar (SAS) system forms a large virtual synthetic array, improving the azimuth resolution. With a constant and high azimuth resolution, it turns to be potential for the image mosaic based on image features. Thus a SAS image mosaic method based on image features is proposed, which uses the SIFT operator to extract the interest points and uses the RANSAC algorithm to do finely matching. The method was applied to the real SAS image, indicating this effective solution.

Keywords: SAS, Image Mosaic, SIFT, RANSAC
1 INTRODUCTION

Side scan sonar is one of the most important means in sea-bed imaging, which may produce stripmap image. The width of image with a single scanning is generally small, thus the images need to be mosaicked to produce wide swath images[1]. For the traditional real aperture side scan sonar, the geography information is applied to the mosaic rather than the image features, which is for the reason that the azimuth linear resolution is in direct proportion to the target distance. Then the azimuth resolution declines as range increase and the inhomogeneity induces the difficulty of extracting the image features.

In synthetic aperture sonar (SAS) system, the platform moves along a line while transmitting and receiving acoustic signals. Combing a number of acoustic pings, the system forms a large virtual synthetic array. Reorganizing the data from all the pings coherently, a synthetic aperture image with improved azimuth resolution is produced[2,3,4]. With a constant and much higher azimuth resolution[5], the SAS results turn to be potential for the image mosaic based on image features, comparing to real aperture side scan sonar.

Scale invariant feature transform (SIFT) operator is a novel method of describing the image feature points, proposed by David G. Lowe in 2004 who summed up the existing feature detection methods using invariant technology[6,7]. The operator is invariant to image translation, rotation and scaling, partially invariant to illumination changes. Therefore it is used widely in object recognition, image mosaic, navigation, 3D modeling, etc.

Random sample consensus (RANSAC) algorithm, suggested by Fischler and Bolles in 1981, is a classic method to estimate parameters from a set of observed samples which might contain many outliers[8].

Finely matching, which employs the SIFT operator and the RANSAC algorithm to calculate feature points, is studied in the paper.

2 FEATURE-BASED SAS IMAGE MOSAIC FLOW

The flow of SAS images mosaic based on image features is shown in Fig1.

![Fig1 SAS Image Mosaic Flow Based on Image Features](image)

2.1 Pre-Processing

The SAS image’s pre-processing consists of slant range correction, speckle reduction and brightness equalization, etc.

The direct striped SAS image locates in a coordinate whose directions are slant range and azimuth separately. Thus it is necessary to transform the image from slant range to horizontal range. The equation for the slant range length \( r \) referred to the target horizontal range length \( x \) is

\[
    r = \sqrt{x^2 + h^2} - r_{mn}
\]
where $h$ is the height of the sonar to the seabed, $r_{\text{min}}$ is the minimum range to be imaged. The slant range is corrected by means of the image data interpolation using equation (1) in x direction.

Speckle is a connatural noise in synthetic aperture images for their coherently processing. There are two categories of despecklisation methods. The first is multi-look processing [9], and the second is filtering [10].

SAS image intensity is range-dependant because of transmission loss of underwater sound, which includes spreading, absorption and scattering. The spreading loss obeys the inverse-square rule, while the absorption and scattering losses obey the exponential loss rule [11]. In order to compensate these losses, time-varying gain (TVG) is usually applied to the sonar echo data. However, TVG could not compensate for the loss perfectly usually for the altitude variations and other diversifications of conditions. According to the attenuation principle, the signal intensity could be expressed by the equation as follows.

$$\lg I = \lg I_0 - 2 \lg (r + r_{\text{min}}) - 2 \alpha \cdot \lg e \cdot (r + r_{\text{min}})$$  \hspace{1cm} (2)

Wherein $I_0$ is the intensity at the reference point located 1 meter from the source ($10 \lg I_0$ is the source level). $I$ is the intensity at a distant point located $r$, $\alpha$ is the attenuation coefficient which includes the absorption coefficient and scattering coefficient. Equation (2) could be deformed into

$$\lg I = a \lg (r + b) + c \cdot r + d$$  \hspace{1cm} (3)

Wherein $a, b, c, d$ are constants. Thus the residual TVG accords to the equation (3). After the estimation of $a, b, c, d$ using the image data, the intensity could be equipoised then.

2.2 Image Feature Extracting

The features usually used in image registration consist of edge features[12,13], point features[14,15,16]and area features, etc. The SIFT feature points descriptor represents of the aim of this paper.

2.3 Feature Matching

Feature matching means comparing features in different images and obtaining a couple of features which indicates the same point. There are two steps in the stage, coarse matching based on Euclidean Distance and the fine matching based on RANSAC.

In the coarse matching step, for one feature point on the reference image (also called the fixed image), the closest and the second closest feature points in the input image which is to be transformed are firstly found using Euclidean Distance. If the ratio of the closest distance to the second is less than a threshold, the closest feature point and the point on the reference are considered as a matching pair.

Due to the noise and other reasons, there are usually many false pairs in the coarse matching ones, which might badly affect the estimation of transformation matrix. Thus the fine matching is essential, in which the RANSAC algorithm is applied.

2.4 Transformation Matrix Determining

After the matching of feature points, the transformation matrix could be determined [17]. The transformation between the reference image and the input image in the affine transforms accords to the formula as follow.

$$\begin{bmatrix} wx' \\ wy' \\ w' \end{bmatrix} = \begin{bmatrix} h_1 & h_2 & h_3 \\ h_4 & h_5 & h_6 \\ h_7 & h_8 & 1 \end{bmatrix} \begin{bmatrix} x \\ y \\ 1 \end{bmatrix}$$  \hspace{1cm} (4)

Supposing that $x_i = (x_i, y_i, 1)^T$ is the homogeneous coordinate of the point on the reference image, while $x'_i = (x'_i, y'_i, w'_i)^T$ is on the input image, the transformation equation could be written as follow then.

$$x_i' = Hx_i$$  \hspace{1cm} (5)

Wherein $H$ is the transformation matrix.
For that the transformation matrix has eight degrees of freedom, it needs 4 points matching pairs to calculate the matrix. When the number of matching pairs is more than four, the solution of the equation is generally not strictly determined. Therefore the singular value decomposition (SVD) algorithm is applied in the paper.

2.5 Interpolation and Blending

The transformation converts the integer coordinates to the real coordinates, which could not be displayed directly. Thus the interpolation is applied in order to get pixel in the integer coordinates. Commonly used methods consist of nearest neighbour interpolation, bilinear interpolation, and bicubic interpolation.

On the overlapped areas of two or more SAS images, there are usually more than one pixel values on each coordinate, which are supposed to be fused into one pixel value. Common fusion methods consist of minimum method, maximum method, arithmetic mean, weighted mean, Gaussian fusion and wavelet fusion.

3 EXTRACTING FEATURE POINTS BASED ON SIFT OPERATOR

3.1 Basic Flow of SIFT Operator

The scale-invariant feature means that for the same object in different pictures regardless of their scale, the same feature points could be extracted using SIFT algorithm. The coarse scale could be got by smoothing the picture in fine scale in a given method. The calculation process of SIFT operator is as follows.

(1) Detecting the scale-space extrema

Let the original 2-D image is described as $f(x,y)$. The 2-D Gaussian kernel function is

$$G(x,y,\sigma) = \frac{1}{2\pi\sigma^2} \exp\left(-\frac{x^2+y^2}{2\sigma^2}\right)$$  \hspace{1cm} (6)

The Gaussian space could be produced from the convolution of a Gaussian kernel function with the input image, $f(x,y)$.

$$L(x,y,\sigma) = G(x,y,\sigma) \otimes f(x,y)$$  \hspace{1cm} (7)

The DoG operator is approximately proportional to the difference of Gaussian kernel and could be expressed by the following formula.

$$D(x,y,\sigma) = (G(x,y,k\sigma) - G(x,y,\sigma)) \otimes f(x,y)$$

$$= L(x,y,k\sigma) - L(x,y,\sigma)$$  \hspace{1cm} (8)

The step of calculating the DoG space of image is as follows.

(a) Convolve the reference image with the Gaussian kernels with different scale factors;
(b) Down-sample the reference image by a factor of 2 to get the next octave of scale space.

By repeating the two above steps, the Gaussian images are produced in different octaves and scales. In the same octave, adjacent Gaussian images are subtracted to produce the difference-of-Gaussian (DoG) images.

By comparing a point’s value to its 26 neighbours in 3x3 regions at the current and adjacent scales in the DoG images, the maxima and the minima are detected, which are called the interested points or the keypoints.

(2) Localizing the keypoint

Two types of unstable keypoint are rejected in this step. The first ones are that have low contrast, and the second are localized along an edge. Then the original location of extrema is determined.

The Taylor series for $D(x,y,\sigma)$ at an extrema point A is

$$D(X) = D + \frac{\partial^2 D}{\partial X} X + \frac{1}{2} X^T \frac{\partial^3 D}{\partial X^2} X$$  \hspace{1cm} (9)

Take the derivative of $D(X)$ with respect to x and set it to zero. There is

$$\hat{X} = \frac{\partial^2 D}{\partial X^2}^{-1} \frac{\partial D}{\partial X}$$  \hspace{1cm} (10)

The location of the extrema point is determined by interpolating the above formula.
Substitute $\tilde{x}$ into $D(X)$, giving $D(\tilde{x})$. Points of low contrast could be rejected by comparing the absolute value of $D(\tilde{x})$ and the preset threshold.

The keypoints on an edge are rejected mainly by using the normal curvature. The point on a non-flat continuous plane has two main directions, whose normal curvatures are the maximum and minimum respectively. For a point on the edge, the curvature which is perpendicular to the edge is the maximum, while the one parallel to the edge is the minimum. Thus if the ratio of the maximum to the minimum is greater than a supposed threshold, then the point is considered to be on an edge and to be rejected.

3) Assigning the Orientation

A local region around the keypoint is given to calculate its gradient and direction. The keypoint's circumference is divided into 36 bins, each of which covers 10 degree. According to the direction of each point in the local region, add its gradient to the 10 degree bin's gradient. In the final gradient histogram, the direction which has the largest gradient is appointed to be the dominant direction of the keypoint. But if another direction has a gradient greater than 80% of the largest one, it is also appointed to be the dominant direction, and in this case the keypoint is considered to have two or more dominant directions [18].

In conclusion, three features of a keypoint are obtained, location, scale and orientation, which brings the invariance of panning, zooming and rotating respectively for SIFT operator.

4) Calculating Feature Point Descriptor

In a local area around the keypoint, e.g. an 8x8 window, the gradient and direction is computed for each point. Then the window is divided into 4 subwindows, and the gradients of the points in each subwindow are accumulated into the gradient histogram, which presents the 8 orientations. Thus a keypoint has a descriptor with a numerical size of 4x8=32.

David suggests that the best results are achieved when choosing a 4x4 subwindows of histograms. Then each keypoint has a descriptor with a size of 4x4x8=128 elements.

3.2 Lake Trial Image Processing

Fig 2 shows a SAS image of seabed in Qiandao Lake in Zhejiang Province, China, with a range length of 150 meters and an azimuth length of 85 meters.

![Fig 2 A lake trial SAS image](image)

Fig 3 shows the 1st scale Gaussian space and the DoG space respectively.

![Fig 3 1st scale Gaussian space (left) and the DoG space (right) of the lake trial SAS image](image)
Fig 4 shows the keypoints’ gradient and orientation, where the length of arrow denotes the gradient and the direction of arrow denotes the orientation of keypoint.

![Fig 4 Grades and directions of SIFT interested points of lake trial SAS image](image)

4 FINE MATCHING BASED ON RANSAC

4.1 Basic Flow of RANSAC Method

RANSAC is an iterative method using the data set’s internal relations to reject the mismatches and to estimate a more accurate model. For the SAS image matching, the data set is the feature point matches via the coarse matching and the model is the transformation matrix. The steps are as follows.

(a). Select $s$ samples randomly from the data set $S$ and estimate the maybe model using the selected samples.

(b). All other samples are tested against the maybe model to see if the sample fits the model with an error smaller than $d_a$. The fitted sample is considered to be an inlier, and the unfitted sample outlier. The inliers form a consensus set $S_i$, which is a subset of $S$.

(c). If the number of elements in the consensus set $S_i$ is bigger than a threshold $N_a$, then estimate the model using all the samples in consensus set $S_i$ and end the process.

(d). But if the number of elements in $S_i$ is less than $N_a$, then select a new subset of samples, and repeat the above process.

(e). Suppose another number $N$. If the random selections is more than $N$ but each selection could not satisfy the condition that the elements are more than $N_a$, end the process also. Then use the $S_i$ which has the largest number of inlier elements to estimate the model.

4.2 Lake trial Images Processing

Fig 5 shows two real SAS images with overlapped area. Fig 6 is the coarse matching result which has 72 keypoint matches. Fig 7 shows the result after the fine matching using RANSAC method, which has 28 matches. From the comparison it could be seen that the mismatches are rejected well by RANSAC method.

![Fig 5 Two SAS Images with Overlapped Area](image)

![Fig 6 Coarse Matching Result (72 Matches)](image)

![Fig 7 Fine Matching Result (28 Matches)](image)

5 RESULTS

Fig 8 shows the mosaic result from the two SAS images shown in Fig 5. Each original image has a range length of 150 meters, while the mosaic image 220 meters.
CONCLUSION

Comparing to the real aperture Side Scan Sonar, the SAS is advantageous because of constant azimuth resolution, which makes the mosaic based on image features possible. The paper proposes a method of SAS image mosaic using point features, whose keys are extracting the keypoints based on SIFT operator and fine matching based on RANSAC method. The processing results of SAS images of Qiandao Lake seabed validate the proposed method.

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RESEARCH ON UNDERWATER TARGET DETECTION BASED ON SEAFLOOR PHYSIOGNOMY-MATCHING OF SIDE SCAN SONAR IMAGERY

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Abstract: This paper proposed an underwater target detection method based on seafloor physiognomy-matching of side scan sonar imagery, which uses the seafloor imagery physiognomy-matching result between the historical sonar imagery and the measured one to detect and confirm the underwater target. When the side scan sonar scans the same region of seafloor at different times, the sonar imageries are different in geometry distorting, compressing and stretching changes due to the differences of the height of the tow body, the orientation of the navigation and the gesture. These changes could result in the deviating or disappearing of the target position on the sonar imagery, but the seafloor physiognomy feature would not disappear. This article preprocessed the measured sonar imagery with noise reduction, background equilibration and geometrical correction, and then used the Sift matching algorithm, a invariant characteristic method, to extract the key terrain feature of the sonar imagery. With the matching result of the terrain feature, the target could be detected. The terrain feature matching algorithm of the side scan sonar imagery has a significant meaning to the harbor safety and underwater target confirming.

Keywords: Side-scan sonar, seafloor physiognomy-matching, target-detection
1. INTRODUCTION:

When the side scan sonar scans the same region of seafloor at different times, the sonar imageries are different in geometry distorting, compressing and stretching changes due to the differences of the height of the tow body, the orientation of the navigation and the gesture. Therefore, an exact matching of the variational sonar imageries in different voyages is the key to this research. In order to solve this problem, this study adopts the Sift matching algorithm with same characteristics. Sift matching algorithm is a kind of local invariant feature points extracting method making rotating, level scaling and brightness changing keep the same; and also keeping the stability of the perspective transformation, affine transformation and noise.

This research solves the problem of the target detection based on the seafloor physiognomy-matching. Altogether there are four processes included:

1) Imagery Pre-processing. By imagery noise reduction, edge sharpening and background equalization preprocessing, showing a more prominent geomorphologic features of imageries.

2) Imagery Matching. By the extraction of geomorphologic features of imageries, judging the compatibility of different imagery features to complete the match of the same geographic point to different imageries.

3) Geometric Correction. By geometric corrections including compression, stretching and rotating, making two imageries under different conditions of sweeping survey of side-scan sonar imagery into the same coordinate system.

4) Target Detection. To increase the target points by comparing historical sonar imageries.

2. SIGNAL PROCESSING:

2.1 Imagery Pre-processing
Imagery pre-processing includes sonar imagery denoising and imagery enhancement. Imagery denoising adopts the median filtering method by which on the one hand, impulse noises can be filtered out from the imagery; on the other hand, the marginal information can be well maintained. Imagery enhancement uses histogram equalization.

2.2 Imagery Matching

Two feature points are employed in the current matching of the historical sonar imagery and the measured one. The first one is the prominent seafloor physiognomy features; for example submarine plateaus and mountain ranges, etc. The second one is artificial targets with some certain characteristics; like underwater shipwrecks, submarine anchor chains, etc. These targets have significant features, which are in a stable state and uneasy to change. However, the measured sonar imagery in the same geographical point shows discrepancies under different sweeping surveys. To ensure matching results, this project uses the image matching technology based on the Sift matching algorithm.

There are two phases in the Sift matching algorithm. One is the production of SIFT features. During this phase, some irrelevant vectors like size scaling, rotating, brightness changing are extracted from those imageries to be matched. The other phase is the matching of SIFT characteristic vectors. In actual sample, samples are extracted in the neighborhood windows centered by the key point; histograms are used to count the gradient direction of the field pixel. The gradient histogram ranges from 0-360 degrees; for each 10 degrees one column, with a total of 36 posts. The histogram peak represents the main direction of the key points in the neighborhood gradient, as key points in this direction. Figure 2 is the gradient histogram by using 7 columns as the key point directions.

![Fig.2: Main direction of the gradient histogram](image)

Within the gradient histogram, while there is another peak of 80% energy of the main peak value, one key point possibly can be assigned in more than one direction (a principal direction and another auxiliary directions), which can enhance the robustness of the matching.

So far, the key points of imageries have been finished detected. Each key point has three informations: locations, scalings and directions, from which a SIFT figure region can be determined, and producing SIFT figure vectors.

Firstly rotating the coordinate for the direction of the key point to make sure the swing is not a transsexual.
Fig. 3: Eigenvectors from neighborhood gradient information of key points

Secondly, taking the 8 * 8 window in the centre of the key point, the black point at the left part in figure 5 is the current location of the key point. Each cell represents a pixel in a criterion scale space where the key point locates nearby. The direction of the arrow represents the pixel’s gradient direction; the length shows the gradient mode value; the blue circle in the above figure is the scope of the Gaussian weighted (closer to the pixel of the key point, the more information of the gradient direction). And then calculate gradient histogram of 8 directions at each 4 * 4 small blocks, drawing added value for each gradient direction to form a seed spot, as showed at the right part of the above figure. In this Figure, a key point is consist of 4 seed spots by 2 * 2 one. Each seed spot has 8 directions vector information. The thought of neighborhood directive information union enhances the ability to resist noise algorithm; at the same time providing a good fault tolerance for the feature matching contained with position error.

During the actual calculation process, in order to strengthen the matching robustness, every 4*4 with totally 16 seed spots are used for each key point; in this way, each key point can produce 128 data and finally form 128D SIFT feature vectors. Therefore, influences of geometry deformation factors such as scale changing and rotation for SIFT feature vectors have been removed.

After the two images’ SIFT feature vectors produced, we use Euclidean distance of the key point’s feature vectors as the criticalities of two images’ key point to determine metrics. Take one key point in figure 1, finding the nearest two key points to the Euclidean distance in figure 2. Within these two points, if the result of the closest distance divided by the closer distance is less than the proportion threshold value, this pair of matching points can be accepted; while if the proportion threshold value is reduced, the number of SIFT matching points will be decreased, but more stable.

The following figures are results of seafloor physiognomy-matching of two side scan sonar imagery.

(a) Results of seafloor physiognomy-matching
2.3 Imagery geometric Correction

Geometric correction method uses the method of remote sensing images for reference. Geometric correction includes two processes: coordinate transformation and re-sampling, whose simple process is shown in the following figure:

*Fig. 5: Processes of geometric correction*
2.4 Target Detection

The confirmation of suspected targets is mainly through the analysis of differences between the measured sonar imagery and the historical one. Then new targets can be detected. When distinguish sonar imageries, target detection method consists of the spot test and the shadow test.

Detection results are as following:

(a) The historical sonar imagery
(b) Testing sonar imagery (targets)
(c) Result of target detection

Fig.7: Results of new targets detection
3. CONCLUSION

We automatch targets based on the matching results of physiognomy features, acquiring target sonar imagery through high resolution, establishing sonar imagery database by imagery pre-processing and sonar imagery joining. And then write the obtaining information into the database. As for new sonar imageries, make suspected targets and typically featured physiognomy sonar imageries be inputted into sonar imagery database and process physiognomy feature SIFT matching. Target matching can be realized by the similarity, location and discrepancy among results of matching, suspected targets and confirming targets. The experiment shows that this method can efficiently improve the ability of an underwater target detection of the side scan sonar imagery.
DESIGN AND IMPLEMENTATION OF A REAL-TIME 3-D
IMAGING SONAR SIGNAL PROCESSING SYSTEM ON
TMS320C6678

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Abstract: In this paper, the design and implementation of a real-time 3-D imaging sonar signal processing system on TMS320C6678 are presented. Owing to the large number of channels and beams, the computational load is prohibitive for real-time 3-D imaging sonar system. In this system, a parallel frequency beamforming algorithm on TMS320C6678, which is efficient and computationally advantageous, is proposed to process the signals coming from a 3-D scene placed in far field. And the procedures of sonar signal processing and data transmissions are illustrated.

Keywords: Acoustic imaging, Imaging sonar, 3-D acoustic imaging, TMS320C6678
1. INTRODUCTION

Acoustic imaging systems are widely used in underwater applications, such as underwater construction research, environmental studies, survey of shipwrecks, dredging, and offshore oil detection. Three-dimensional imaging sonar has been one of most important innovations in these underwater applications and plays an important role in underwater acoustic imaging systems for many years. In this paper, the design and implementation of a real-time 3-D imaging sonar signal processing system on TMS320C6678 are presented. And the procedures of sonar signal processing and data transmissions are illustrated.

This paper is organized as follows: Section II presents the far-field beamforming method for underwater 3-D sonar imaging which is suitable for multi-core DSP system. Section III presents the design and implementation of 3-D sonar imaging processing system based on TMS320C6678. Section IV demonstrates the test results of the system in water tank. Finally, some conclusions are drawn in section V.

2. THE FAR-FIELD BEAMFORMING FOR UNDERWATER 3-D SONAR IMAGING

Let us consider a planar array placed on the plane \( xy \) consisting of \( M \times N \) sensors. The sensor, which is identified by indices \((m,n)\), is placed at position \((x_m,y_n)\) and the unit vector of steering direction \( u \) can be presented as

\[
u = (u_x, u_y, u_z)
\]

To decrease the computational burden and the hardware requirements of digital beamforming, Fig. 1 shows the geometry and notation proposed in [1] for a planar array, where \( \theta_a \) and \( \theta_e \) are the azimuth and elevation steering angles, respectively. The unit vector of steering direction \( u \) can be represented as

\[
u = \left( \sin \theta_a \sin \theta_e, \sqrt{\cos^2 \theta_a - \sin^2 \theta_e} \right)
\]

\[\text{Fig.1: Notation and geometry for a 2-D array}\]

Let us consider the \( P \times Q \) beam signals in the azimuth and elevation direction need
to be computed, respectively. The beam signal of steering direction \((\theta_{ap}, \theta_{ep})\) can be expressed as

\[
b(t, \theta_{ap}, \theta_{ep}) = \sum_{m=0}^{M-1} \sum_{n=0}^{N-1} \omega_{mn} s_{mn}(t - \tau(\theta_{ap}, \theta_{ep}, m, n))
\]

where \(s_{mn}(t)\) is the time signal received by the index \((m, n)\) sensor, \(\omega_{mn}\) is the window weight assigned to such a sensor to control the side-lobe level, and \(\theta_{ap}\) and \(\theta_{ep}\) \((1 \leq p \leq P, 1 \leq q \leq Q)\) are the two angles chosen from a set of azimuth angles and a set of elevation angles, respectively. If the time series \(s_{mn}(t)\) resulted from the temporal sampling of the signals gathered by the sensors are segmented into blocks of length \(K\) and transformed into their frequency-domain versions \(S_{mn}(k)\) by Fast Fourier transform (FFT). Then the DFT coefficients \(B(k, \theta_{ap}, \theta_{ep})\) of the beam signal \(b(t, \theta_{ap}, \theta_{ep})\) in far-field region are given by the following expression:

\[
B(k, \theta_{ap}, \theta_{ep}) = \sum_{m=0}^{M-1} \sum_{n=0}^{N-1} \omega_{mn} S_{mn}(k) \times \exp(-j2\pi f_k \tau(\theta_{ap}, \theta_{ep}, m, n))
\]

\[
= \sum_{m=0}^{M-1} \sum_{n=0}^{N-1} \omega_{mn} S_{mn}(k) \times \exp\left(-j2\pi f_k \left(\frac{md \sin \theta_{ap} + nd \sin \theta_{ep}}{c}\right)\right)
\]

where

\[
f_k = kf_s / K
\]

Let us call \(f_s\) the sampling frequency for each temporal signal. As 3-D real-time acoustical imaging system require the spatial sampling of the acoustic field on planar array and the calculation of a large number of beam signals, the cost of hardware and the computational burden in processing the signals turn out to be very expensive. A large number beam signals must be computed to generate a 3-D image by digital beamforming. Equation(6) expresses \(B(k, \theta_{ap}, \theta_{ep})\) as phase-shift beamforming in quadrature, which can be to used to compute the beam signals necessary for image formation. The phase-shift beamforming can be implemented with FFT method which results in a very fast computation of all necessary beam signals. Let us calculate the two dimensional FFT on the sensor indexes as follows:
\[ B(k, f_1, f_2) = \text{FFT}\{S(k)\}_{M \times N} \]
\[ = \sum_{m=0}^{M-1} \sum_{n=0}^{N-1} S_{mn}(k) \exp(-j2\pi f_1 m / M) \exp(-j2\pi f_2 n / N) \] (7)

Let us determine the relations linking each pair \((f_1, f_2)\) to \((\theta_{ap}, \theta_{ep})\) by the following equation:
\[ |B(k, f_1, f_2)| = |B(k, \theta_{ap}, \theta_{ep})| \] (8)

If we denote the distance between two adjacent sensors along the \(x\) and \(y\) axes using \(d\), then the relations are given by
\[
\begin{align*}
\theta_{ap} &= \arcsin \frac{f_2 c}{d M f_k} & 0 \leq f_1 \leq M / 2 - 1 \\
\theta_{ep} &= \arcsin \frac{(f_1 - M) c}{d M f_k} & M / 2 \leq f_1 \leq M - 1 \\
\theta_{ap} &= \arcsin \frac{f_2 c}{d N f_k} & 0 \leq f_2 \leq N / 2 - 1 \\
\theta_{ep} &= \arcsin \frac{(f_2 - N) c}{d N f_k} & N / 2 \leq f_2 \leq N - 1
\end{align*}
\] (9)

3. DESIGN AND IMPLEMENTATION ON TMS320C6678

The system consists of one TMS320C6678 DSP which is highest-performance fixed/float-point DSP based on KeyStone multicore architecture. It uses the interface of SRIO to connect with FPGA and transmits the beam signals to PC by Ethernet. The structure of the system for 3-D imaging sonar signal processing is shown in Fig.2.

Fig.3: The structure of the system for 3-D imaging sonar signal processing

As shown in Fig.2 the system consists of three main functional parts, pre-processing part, imaging signal processing part and imaging data transmitting part. The SRIO in the system is very important because it brings much greater
bandwidth than the traditional data transmitting way. The DDR3 is mainly used for data-storage and ping-pong processing while the shared memory is used to store the shared parameters for the signal processing and the program. The raw data coming from FPGA is stored in the DDR3. The objective of pre-processing part is to apply FFT beamforming to real data acquired from sparse planar arrays and allocate data after pre-processing to each core of imaging signal processing part.

4. TEST BY A WATER TANK EXPERIENCE

In order to assess the validity of the proposed system, a water tank experiment is presented. The dimensions of the water tank are about 10m length, 5m width and 5m depth. The test target is an iron bar of height=2m and radius=3cm. The experiment scene here is presented and shown in Fig.5. The receiving array is a planar array consisting of $48 \times 48$ sensors using $d = 0.4cm$ to denote the distance between two adjacent sensors. There is an acoustic transmitter insonifying all the volume of interest by a pulse $3.3 \mu sec$ long with a carrier frequency of 350 KHz.

![Fig.5: The experiment scene](image)

While scanning the whole area, the 3-D imaging sonar system acquired 2048 samples by ping corresponding to all the transducer’s positions. A sampling frequency of 2MHz led to an acquisition length of about 50 m. The obtained images from the 3-D imaging sonar signal processing system shown in Fig.6. Finally, Fig.7 shows the three dimension visualization of a iron bar used for experiment in water tank.

![Fig.6: Strata representation of imaging data. Strata thickness:0.024 cm. The first stratum starts at 4m.](image)
5. CONCLUSIONS

In this paper, the far-field beamforming method for underwater 3-D sonar imaging, which is efficient and computationally advantageous based on multi-core DSP system, is illustrated. The design and the implantation of a real-time 3-D imaging sonar signal processing system on TMS320C6678 are presented. This system consists of pre-processing, imaging signal processing and imaging data transmitting. The results obtained from the signal processing system have shown that it is able to produce imaging data efficiently.

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NUMERICAL ANALYSIS FOR AMBIENT NOISE IMAGING WITH ACOUSTIC LENS: TARGET DETECTION AROUND THE BARGE MOORED IN UCHIURA BAY

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Abstract: We designed and made a prototype system for Ambient Noise Imaging (ANI) with an aspherical lens. The system was deployed through the barge ‘OKI SEATEC II’ moored in Uchiura Bay on November 8-13, 2010. We successfully detected some silent targets under dominant snapping shrimp noises. The noise spatial distribution was also observed using two tetrahedron arrays. Now, we are planning a second sea trial at the same site in November of 2014. The main objective is to verify whether the target can be successfully imaged under specific conditions governing the direction of the imaging system and the noise distributions. In this study, we conducted various numerical simulations of sound propagation using the finite difference time domain method. Arranging many noise sources in a spatial distribution similar to that observed in 2010, the sound pressures received at the focal array were calculated when the field of view contained an area in which the noise sources were not present. The sound pressures corresponding to the on-target directions were greater than those for off-target directions, because only the receivers for the on-target directions received the target scatterings. The sound pressures were also calculated when the field of view included an area where the noise sources were present. The on-target pressures were weaker than the off-target pressures because the noises directly received by the off-target receivers were much greater than the pressures of the target scatterings received by the on-target receivers. These findings will help determine the arrangement of the prototype system and targets in the planned sea trial.

Keywords: Ambient noise imaging, Acoustic lens, Numerical Analysis
1. INTRODUCTION

The radical idea of viewing ambient noise as a sound source rather than a hindrance was developed by Buckingham et al. The detection method based on this idea makes use of neither passive nor active sonar, and is often called ambient noise imaging (ANI) [1]. Some research groups have built experimental systems to perform ANI. Epifanio et al. developed the Acoustic Daylight Ocean Noise Imaging System consisting of a 3-m-diameter spherical reflector with an array of 126 hydrophones attached to the focal surface [2]. Venugopalan et al. built the Remotely Operated Mobile Ambient Noise Imaging System (ROMANIS) consisting of a 2-D sparse array of 504 hydrophones fully populating a 1.44-m circular aperture [3]. Both systems succeeded in detecting silent target objects under dominant snapping shrimp noises. Recently, Chitre et al. rebuilt the ROMANIS and created stable target images. They also successfully estimated the target range using noise source positions at the same time [4].

Various sound pressure fields focused by lenses constructed for an ANI system were analysed in our previous studies [4-7]. Recently, we designed and built an aspherical lens with an aperture diameter of 1.0 m with which to develop a prototype ANI system. In 2010, a sea trial of ANI was conducted with the prototype system, which was constructed by mounting a hydrophone array on the image surface of this lens, in order to measure its directional resolution. It was verified that this acoustic lens realizes directional resolution with a beam width of 1 degree at the centre frequency of 120 kHz over a field of view from −7 to +7 degrees [8]. On November 8-13, 2010, another sea trial of silent target detection was conducted under only background noise in Uchiura Bay, Japan. Many transient sounds were detected by hydrophones arranged on each image point. Then, we classified the received transients roughly into directly received noises and target scatterings. A classification method was proposed to extract only transients classified as target scatterings. Finally, it was verified that the power spectrum density levels of the on-target directions were greater than those of the off-target directions in the higher frequency band over 60 kHz using the data classified as target scatterings. Thus, we succeeded in detecting the silent targets under ocean background noise generated mainly by snapping shrimp [9].

Now, we are planning a second sea trial at the same site in November of 2014. The main objective is to verify whether the target can be successfully imaged under various conditions involving the direction of the imaging system and the noise distributions. In the 2010 sea trial, a pair of tetrahedral arrays was used near the prototype ANI system in order to observe the spatial distribution of the noise sources. Four hydrophones were mounted on each tetrahedron with a separation of 1 m on each array. Figure 1 shows the estimated source positions for the vertical plane around the barge. We can see that a portion of the noise sources is concentrated at the bottom of the barge around the sea surface, and the remaining noise sources are spread around the sea bottom [10, 11]. In this study, we conducted various numerical simulations of sound propagation using the finite difference time domain (FDTD) method. Arranging many noise sources in a spatial distribution similar to that used in the 2010 sea trial, the sound pressures received at the focal array were calculated when the field of view was composed of an area where the noise sources were not present. In this case, the prototype system was arranged nearly horizontally as shown in Fig. 1. In addition the sound pressures were calculated when the field of view included the area containing the noise sources. The prototype system was then arranged vertically and its viewing direction was toward the sea bottom.
2. NUMERICAL SIMULATION

Figure 2 shows the arrangement of the analysis domain using the 2-D FDTD method including the attenuation in the media, which was developed by Iijima et al [12]. Here, we considered a spatial distribution of noise sources similar to that shown in Fig. 1. The rigid barge was arranged on the sea surface as shown in Fig. 2(a). There were 90 point sources at the bottom of the barge and 100 point sources on the sea bed. Each source independently radiated Gaussian noise in the limited band from 20 kHz to 120 kHz. The sound speeds and densities were 1500 m/s and 1000 kg/m$^3$ in the water and the absorption layer, and 1800 m/s and 1800 kg/m$^3$ in the sea bed, respectively. The attenuation constant in the water was 0.0 dB/$\lambda$, because attenuation in the water was neglected. The attenuation constants were 5.0 dB/$\lambda$ in the absorption layer and 1.0 dB/$\lambda$ in the sea bed. Mur’s first-order absorbing boundaries were applied to the left, right and bottom outer boundaries of the analysis domain in order to eliminate the reflection wave from those boundaries. A part of the top outer boundary of the analysis domain was set up with a “pressure release” condition on the sea surface, and the other part was set up with a “no normal velocity” condition in the rigid barge. The water depth and the size of the barge were 30 m. The range between the imaging system and the rigid target was 15 m. The size of the target was 1 m. We then considered two conditions for the directions of the imaging system. The first condition (Case A) was nearly horizontal, and the field of view (FOV) included an area where the noise sources were not present. The second condition (Case B) was nearly...
vertical, and the field of view included an area where the noise sources were present. The imaging system then was directed toward the sea bed. In both cases, the imaging system and target were placed so that they are facing each other, and the target was arranged in the centre of the field of view. As shown in Fig. 2(b), the imaging system was constructed with the aspherical acoustic lens and receivers. The aperture diameter of the lens was 1.0 m, and the centre thickness was 0.01 m as designed in Ref. [8]. To image the target, 15 receivers were arranged on the positions of the image points. These corresponded to the incidence angles of $-7$, $-6$, $...$, $+7$ degrees. The time series of the sound pressure on each receiver point was recorded over all time-steps of the analysis. The rigid shields and absorption layers were mounted on both sides of the lens and on the rear of the receivers, so that the noises from the point sources were not directly within the focusing area. The sound speed, density and attenuation constant in the lens were 2670 m/s, 1200 kg/m$^3$, and 1.0 dB/λ, respectively.

Fig. 2: Analysis domain for numerical simulation.
3. ANALYSIS RESULTS AND CONCLUSIONS

Figure 3 shows the relative received power level vs. the angle of incidence for the imaging system. The relative power level, which is the band level from 40 kHz to 120 kHz calculated by the power spectrum of the sound pressure at each receiver point, was normalized by the maximum value. In the conditions used in Case A, the relative powers corresponding with the on-target directions from $-3$ to $3$ degrees were greater than those for the off-target directions because only the receivers for the on-target directions received the target scatterings. However, the maximum was formed at $-2$ degrees of the angle of incidence. This bias suggests that the target scatterings were mainly generated by the noise sources at the bottom of the barge. On the other hand, the on-target powers from $-3$ to $3$ degrees were weaker than the off-target pressures, because the noises directly received by the off-target receivers were much greater than the pressures of the target scatterings received by the on-target receivers in the conditions used in Case B. This suggests that the target was silhouetted against the noises arranged on the sea bed. The contrasts of the relative power were over 20 dB in Case A and about 7 dB in Case B.

The analysis results showed that the relative received power vs. the angle of incidence drastically changed based on the arrangement of the imaging system, the target, and the spatial distribution of the noise sources. Future studies like the present analysis are needed to survey other arrangements. Such studies will help determine the arrangement of the prototype imaging system and targets to be used in the planned sea trial.

\begin{figure}[h]
\centering
\includegraphics[width=\textwidth]{example.png}
\caption{Relative received power level vs. angle of incidence.}
\end{figure}

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Session 2

Acoustic Monitoring of Marine Mammals

Organizers: Purnima Ratilal and Adam Zielinski
PASSIVE ACOUSTIC MONITORING OF HUMPBACK WHALES IN EXMOUTH GULF USING A SPARSE ARRAY OF DIFAR SENSORS

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Abstract: Exmouth Gulf and the coastal shelf north of it in Western Australia are known as winter resting grounds for the Western Australian population of humpback whales that visit this area yearly from early June to late October. The marine environment over this part of Australia’s continental shelf can potentially be affected by intense offshore operations that currently take place as a result of oil and gas exploration, the development of already discovered fields and associated port infrastructure. Effective means are needed to prevent physiological and behavioural impacts of such operations on humpback whales, including collisions with vessels and the effect of man-made underwater noise. To examine the feasibility and assess the expected capability of a passive acoustic monitoring system to detect and localise humpback whales approaching offshore operation sites, ports and frequently used vessel corridors, a sparse array of DIFAR sensors was deployed over the shelf off Onslow and then in the gulf. The array consisted of three to six drifting sonobuoys deployed from about 1 km to nearly 15 km apart. Acoustic monitoring was accompanied by visual sightings. The detection range and localisation accuracy were determined using bearing data from the DIFAR sensors. Some new types of sound from humpback whales were recorded. It was also observed that pods of cow and calf stayed silent most of the time; however, they were often escorted by vocalising male whales. The research was commissioned by Chevron who funded the work through the Western Australian Energy Research Alliance, as part of the environmental monitoring program for the Wheatstone LNG project.

Keywords: humpback whale, passive acoustic detection, localization
1. INTRODUCTION

Intensive developments of the North West (NW) shelf of Western Australia (WA) related to offshore gas and oil exploration and production the potential to affect the marine fauna inhabiting this region of the ocean. The WA population of humpback whales is one of the major concerns associated with this activity. Every year humpback whales populate extensive areas over the NW shelf during austral winter and spring before their summer migration to feeding grounds in the Southern Ocean. Moreover, Exmouth Gulf and the coastal zone north of it are known as a resting area of humpback cows and calves in WA [1]. The current offshore developments and a substantial intensification of shipping expected in the coming decades may result in direct or indirect impacts on the humpback whale population, e.g. through an increase in the emission of man-made underwater noise or even by direct strikes with vessels. A comprehensive monitoring program is needed to prevent undesirable effects on the WA humpback whale population.

Underwater Passive Acoustic Monitoring (PAM) is becoming an increasingly common approach to study marine fauna and assess potential impacts of offshore industrial activities on the marine life. PAM is relatively cost-effective and, in contrast to visual observations, provides 24-hour monitoring of vocalising marine animals almost independently of weather conditions. PAM systems have been effectively used to monitor the presence and movement of marine mammals, such as whales and dolphins, in the area of interest [e.g. 2].

The Wheatstone Project conducted by Chevron Australia requires several offshore operations on the continental shelf north of Exmouth Gulf which may have an impact on humpback whales populating this area during several months every year. These operations are shipping channel and pipeline trench dredging followed by regular shipping. A feasibility study, including a desktop analysis and a field trial, was commissioned by Chevron to examine the efficiency of PAM to detect the presence of humpback whales in the operation areas and localise and track individual whales in order to prevent potential impacts on them.

This paper describes methods for acoustic detection and localisation of humpback whales and presents some results of the experimental test in the area to be monitored.

2. METHOD

2.1. Experimental measurements

To assess the potential of passive acoustic monitoring of humpback whales, a sea trial was undertaken in the coastal shelf area off Onslow and in Exmouth Gulf (Fig. 1). Five passive acoustic whale monitoring system’s (PAWMS) buoys with DIFAR sensors attached to subsurface legs of 2.5 m long were moored in the northern trial area at the locations shown in Fig. 1. Freely drifting SSQ53F DIFAR sonobuoys were used in the Gulf. All acoustic buoys were deployed from R/V Whale Song which had a PASOR1 receiver on board to record RF signals and GPS data from up to fourteen different sensors at the same time. Few humpback whales were detected and localised by the PAWMS

1 Portable Acoustic Sonobuoy Range, developed by L-3 Oceania
During the first day of the trial, the wind speed increased considerably with frequent gusts up to 30 knots, which resulted in a much rougher sea state and consequently a much higher background noise level at the DIFAR sensors. Moreover, humpback whales moved away from the monitored area most likely due to weather change. A decision was made to move into the Gulf where the wind was noticeably lower and the number of whales was expected to be significantly larger. Four to six DIFAR sonobuoys deployed from the vessel were recording sea noise in the monitored area at the same time. Three to six individual whales were simultaneously acoustically observed from the sonobuoys. All results discussed in Section 3 were obtained from the PAM data collected in the Gulf.

![Location of two PAM trial areas on the northwest shelf of Western Australia (dashed ellipses). Locations of the moored PAWMS buoys are shown by red dots.](image)

### 2.2. Whale detection

Designing a universal and reliable automatic detector of humpback whale sounds is a challenging problem. In contrast to many other whale species that produce a limited number of stereotypic calls, humpback whales have a wide repertoire of different sounds in their songs and social calls. Even similar sounds (units) in a humpback whale song are often made at somewhat different frequencies and have different duration. Moreover, the sound repertoire varies between populations and changes over time. That is why common approaches to automatic detection of whale vocalisation, such as spectrogram correlation, are not effective for humpback whales.

A method for automatic detection of humpback whale calls based on extraction of time-frequency contours in sea noise spectrograms was suggested and examined in [3]. Spectrogram contours of six different frequency modulated calls were extracted using an edge-tracing algorithm commonly used in image processing. The missed detection rate varied from 0 to 84% depending on duration and frequency content of each call at a broadband signal-to-noise ratio (SNR) of 3 dB.

The approach implemented and tested in this study is based on a ridge finding method which is also widely used in image processing [4]. This approach seems more logical than the edge-tracing method, as it extracts time-frequency lines of high intensity in spectrograms rather than contours around the lines. In the case of acoustic detection, the ridges are lines of peak intensity in the spectrogram. Horizontal and sloping lines
correspond to tonal and frequency modulated signals respectively, while vertical lines are broadband impulsive signals. The procedure of finding ridges in an image or spectrogram consists of three steps:

1. The image is scaled by convolution with a Gaussian kernel to reduce noise in it:

\[
L(t, f, \sigma) = g(t, f, \sigma) \ast I(t, f) ,
\]

where \( g(t, f, \sigma) = (1/2\pi\sigma)\exp[-(t^2 + f^2)/2\sigma] \) is the Gaussian kernel, \( I \) is the original spectrogram as a function of time \( t \) and frequency \( f \), and \( \sigma \) is the kernel width. The kernel width is chosen to be similar to the width of the spectrogram lines, which depends on FFT window length and overlap used to calculate the spectrogram.

2. Local directional derivatives up to the second order, \( L_p, L_q, L_{pp}, \) and \( L_{qq} \) are calculated and elements of the spectrogram matrix belonging to the ridges are found based on the following criteria for local ridges:

\[
\begin{align*}
L_p & = 0, \\
L_{pp} & > 0, \\
|L_{pp} - L_{qq}| & \geq 0
\end{align*}
\]

Since results of numerical differentiation always contain errors due to finite differences, zeros in Eq. 2 are replaced with sufficiently small numbers. To keep these numbers unchanged irrespective of the absolute signal intensity, the spectrogram is normalised by the RMS value of its elements.

3. The local strength of each detected ridge is assessed using the following measure:

\[
|L_{pp}^2 - L_{qq}^2|.
\]

The sensitivity of the ridge detector depends primarily on the threshold chosen for this measure.

4. Detected points of a ridge may, in general, have a number of neighbour points in the spectrogram that satisfy the criteria in Eq. 2 and exceed the threshold chosen for the ridge strength. Moreover, two or more ridge lines may overlap in the time domain. To avoid ambiguity in selection of the ridge points that form individual time-frequency lines, all ridge points are firstly clustered in groups where each point has at least one neighbour, including the diagonal ones, from the same group. For each group, the line forming algorithm: (1) takes the first point found in the time domain; (2) selects the point of lowest frequency, if there is more than one point found at the same time; (3) searches for the first connection by selecting the point of highest ridge strength from all neighbours found in the forward direction, which includes low diagonal, forward, high diagonal and upward; and then (4) continues the search/selection operation in the forward direction until the last point of the line is found. The neighbour points not selected at each previous step are excluded from further search.

The resulting time-frequency lines represent different sounds found in the spectrogram. Vertical lines correspond to broadband short impulsive sounds, e.g. snapping shrimp impulsive noise and echo-location clicks by toothed whales. These lines are excluded from further analysis by limiting the minimum duration of detected signals. Long horizontal lines and long lines with a gradually varying slope are usually tonal and frequency modulated man-made noises, e.g. machinery and vessel noises. They are excluded from further consideration by limiting the maximum duration of the signals to be identified as whale calls. Hence the time-frequency lines selected for identification are either tonal/quasi-tonal or frequency modulated sounds of limited duration. Such sounds can be produced by several whale species, e.g. humpback, blue and fin whales, dolphins (whistles), and some fish species.
Supervised or unsupervised learning of the detection algorithm is needed to make it capable of classifying the sounds detected from marine animals. A simplified approach to distinguish humpback whale calls from other sounds was implemented in the current, low-complexity version of the detection algorithm. Firstly, the maximum duration of each sound to be selected was limited to 5 seconds, as the typical units in a humpback song and social sounds are commonly shorter. This eliminates most of the sounds produced by blue whales and man-made noises. Although some units of humpback whale songs contain high-frequency components of several kHz, e.g. high-order harmonics of quasi-tonal sounds, most of the sound energy is concentrated in a lower frequency band. In the detection algorithm used in the experimental test, all sounds above 2 kHz were ignored.

2.3. Localization

The DIFAR sensors provide signals from three channels: one omnidirectional and two perpendicular directional (south-north and west-east), which are multiplexed in a broadband signal transmitted via an RF link from the sonobuys to a multichannel receiver. Once the transmitted signal is de-multiplexed into three channels, bearing to the sound source can be measured using the algorithm described in [5]. According to [6], the distribution of random errors in bearing measurements \( \phi \) can be approximated by a Von Mises distribution, which is analogous to the normal distribution for directional data from 0 to 2\( \pi \):

\[
f(\phi, \phi_0, k) = [2\pi I_0(k)]^{-1} \exp[k \cos(\phi - \phi_0)],
\]

where \( k \) is the concentration parameter, \( \phi_0 \) is the mean bearing angle, and \( I_0 \) is a modified Bessel function of 0-order. The concentration parameter \( k \) can be estimated from the standard deviation \( \delta \) of bearing data using the equations given in Appendix B in [5]. According to sonobuys’s manufacturer, the DIFAR bearing error is \( \pm 10^\circ \), which corresponds to approximately \( 6\delta \). So, the standard deviation \( \delta \) is about 3.3\( ^\circ \).

When two or more DIFAR sonobuys are deployed and receive a signal from the same sound source, the \( x \) and \( y \) position of the source and errors of localisation can be estimated using a maximum likelihood approach described in [7]. In this method, it is assumed that errors of bearing from different sensors are statistically independent and have the same \( \delta \). An error ellipse of chosen confidence of localisation is calculated from the covariance matrix of measurements.

3. RESULTS

3.1. Detection

The SNR of humpback whale calls observed at individual acoustic sensors of the PAWMS array off Onslow was, in general, low due to intense background noise induced primarily by surface waves affecting the moored acoustic buoys. However, almost all humpback calls, identified visually in sea noise spectrograms, were also picked out by the automatic detector. At the same time, no false detections were observed, which was primarily due to the absence of significant man-made noises. Humpback whale calls were automatically detected even when the SNR was about 10 dB in a 1-Hz frequency band.
The detector settings were: FFT window length of 60 ms and 50% overlap to calculate the spectrogram, the minimum ridge strength of 5, and the minimum signal duration of 0.3 s.

In contrast to the PAWMS observations off Onslow, recordings made in Exmouth Gulf contained calls from a number of humpback whales vocalising at the same time. Calls of various types from different individuals were successfully detected in the omnidirectional channel of the DIFAR sensors (Fig. 2). Note that some low-intensity harmonics of quasi-tonal calls were not detected.

![Spectrogram](image.png)

*Fig. 2: Spectrogram of sea noise containing calls from different humpback whales recorded in Exmouth Gulf. Black lines show the time-frequency lines of whale calls extracted by the automatic detection algorithm.*

### 3.2. Localisation

Only a few humpback whale calls were detected at the same time at more than one of the DIFAR sensors of the PAWMS array off Onslow. This was a result of low SNR and, consequently, relatively short detection range, which was estimated to be 4 to 5 km.

In Exmouth Gulf, the background noise level was noticeably lower, which allowed us to detect vocalising whales at distances of more than 10 km. The distance between the sonobuoys deployed in the Gulf varied from about 1 km to nearly 10 km so that calls from the same animal were detected at all receivers deployed at the same time. Figure 3 shows spectrograms of a 20-s fragment of sea noise recordings at four sonobuoys where the bearing angle to sound sources is colour-coded and the sound intensity is illustrated by image brightness. At least four whales vocalising at different locations relative to the sonobuoys can be distinguished in these spectrograms. Three of these whales were close enough to localise their position using the difference in bearing data (Fig. 4).

### 3.3. Humpback whale sounds

Most of the units in whale songs and individual sounds had spectrograms typical for humpback whales of the Western Australian population, with some variations in frequency and its modulation type. However, acoustic recordings accompanied with visual sighting and tracking revealed some new sounds from humpback whales which had never been recorded before in Western Australia. One of these sounds is highlighted by dashed rectangle 2 in the spectrogram shown in Fig. 3. This broadband “mumbling” sound did not contain any distinct frequency component and hence could not be automatically detected by the method discussed in Section 2.2.
Fig. 3: Spectrograms of multiple humpback whale calls received at four DIFAR sonobuoys in Exmouth Gulf. Bearing to the sound source is colour-coded. Dashed rectangles indicate calls from three different whales localised from the sonobuoys’ array (Fig. 4).

Fig. 4: Locations of four DIFAR sonobuoys (blue dots) and three humpback whales localised from the sonobuoys (red dots and dashed ellipses of 95% confidence area)

A pod of caw and calf resting and slowly moving in the monitored area was regularly sighted and tracked in Exmouth Gulf over 3 hours. The distance to this pod from Whale Song varied from about 0.5 km to 2.5 km. An array of four sonobuoys was deployed in the sighting area during this time period. Some sounds arriving from the direction to the pod were observed only at the time around the first sighting. However, no sounds were received from this pod during the rest of the sighting time. Other pods of caw and calf were often escorted by male whales. In all these instances, sounds from these pods were regularly observed and the source was localised. However, it was impossible to identify who of the whale trio made these sounds, as the localization error was much larger than the distance between whales in the pod.
4. CONCLUSIONS

1. A method for automatic detection of humpback whale vocalisation was developed and tested using underwater acoustic recordings made in over the North West shelf in Western Australia. The method is based on tracking high-intensity time-frequency lines in sea noise spectrograms. The test of the detection method showed that it was capable of detecting humpback whale sounds when the SNR was 3 db or higher in a 1-Hz frequency band. Further development of this method is needed to make it capable of distinguishing humpback whale sounds from other acoustic signals with similar time-frequency characteristics.

2. Localisation of vocalising humpback whales was tested using bearing data from a number of DIFAR sonobuoys. The sea trial demonstrated that humpback whales could be acoustically detected and localised at distances up to about 5 km in a noisy marine environment and at more than 10 km at favourable sea conditions.

3. Concurrent visual and acoustic observations revealed some new types of humpback whale calls, which had never been recorded before. These observations also confirmed that pods of cow and calf could stay silent for several hours.

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REFERENCES


ACOUSTIC MONITORING OF MARINE MAMMALS IN THE CHUKCHI SEA — THREE CASE STUDIES

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Abstract: We present an overview of some recent acoustic monitoring studies conducted in the northeastern Chukchi Sea off the north coast of Alaska. Each study exemplifies a different analytical objective and application of underwater instrumentation technology in a remote and challenging natural environment. Three projects, which span a range of approaches and complexities, are examined. In the first two case studies, ambient noise and marine mammal vocalisation data were collected with an array of over 30 autonomous recorders covering an ocean surface of 75,000 square kilometres over the summer months and, with a reduced number of stations, under pack ice during winter. These data were used a) to analyse the movement patterns of several marine mammal species and detect their calls in noise, and b) for research on the spatio-temporal distribution of male bearded seal vocalisations and the seasonal and diel patterns of their acoustic repertoire. In the third case study, two real-time, directional, telemetered acoustic recording systems deployed around walrus haul-outs monitored underwater calls, cross-fixed their original location and, in post-processing, determined their absolute source level and other acoustic properties. Using these examples as a basis, we examine the research possibilities offered by long-term archival or telemetered acoustic recording systems that can operate in extreme environmental conditions. We speculate on future study objectives and new technological developments to help us pursue these opportunities.

Keywords: Passive acoustic monitoring, Chukchi Sea, marine mammals
1. INTRODUCTION

The power of multi-recorder passive acoustic monitoring for measuring temporal and spatial distributions of marine mammals over large areas has been demonstrated by successful research programs spanning multiple years [1]. Acoustic monitoring is effective in nearly all weather conditions and provides a viable solution for species distribution mapping in the Arctic winter when direct monitoring methods such as aerial and vessel-based surveys are hindered or prevented by limited daylight, extremely low temperatures, and presence of ice. Autonomous acoustic recorders enable uninterrupted detection of vocalising marine mammals over long periods. While acoustic monitoring generally cannot track individual animals in the way tagging studies do, it can sample larger fractions of populations at selected sites and yield meaningful temporal distributions of habitat use. Passive acoustic monitoring also eliminates physical contact with animals, thus avoiding the potentially negative effects of animal tagging.

This paper describes three studies of marine mammals conducted in the Chukchi Sea, part of the Alaskan Arctic, by means of passive acoustic monitoring. The first two studies were based on a large-scale monitoring network of more than 30 bottom-mounted autonomous recorders deployed over thousands of square kilometres, the third on a short-range telemetric real-time system that consisted of only two mobile recorders. Considered in combination, these studies convey a sense of the diverse scientific purposes to which acoustic monitoring can be applied and the wide range of complexity of the required infrastructures.

2. CASE STUDIES

2.1. Movement and detectability of marine mammals

A large-scale passive acoustic monitoring (PAM) program was initiated in 2006 in the northeastern Chukchi Sea following the sale of lease blocks to oil companies interested in the potentially hydrocarbon-rich bottom of this region of the Arctic. The main goals of the program were and remain to (1) document seasonal and annual variability of marine mammals’ occurrence, based on acoustic detections of their vocalisations; (2) characterise baseline ambient noise conditions; and (3) characterise sounds produced by oil and gas exploration. These objectives provide a way to detect potential effects of increasing anthropogenic activities on marine mammals, many of which provide traditional subsistence to Alaskan native communities and are considered ecosystem sentinels in an environment increasingly affected by climate change.

Acoustic data collection started in July 2007 and has been virtually uninterrupted since October 2008. The ongoing data collection program has been divided into summer and winter components. Summer data were sampled continuously at 16,000 samples per second and 24-bit resolution using Autonomous Multichannel Acoustic Recorders (AMAR, JASCO Applied Sciences) between late July and mid-October. Up to about forty recorders were deployed each summer, arranged on four lines perpendicular to the coast plus cluster arrays on specific lease areas. Winter data were sampled at 16,384 samples per second and 16-bit resolution using up to eight Autonomous Underwater Recorders for Acoustic Listening, Model 2 (AURAL, Multi-Électronique Ltd). Because of the duration...
of the winter deployment period (October-July), data were only recorded for 40 minute periods every 4 hours (~17% duty cycle). Fig. 1 shows the deployment patterns for the winter and summer periods from 2009 to 2011, covering an ocean surface of about 75,000 square kilometres.

Fig. 1: Acoustic recorder locations for the 2009-2011 winter and summer deployments in the northeastern Chukchi Sea.

Marine mammal vocalisations were detected and classified manually and with automated software [2]. Three species of key interest—bowhead whale (*Balaena mysticetus*), beluga (*Delphinapterus leucas*), and walrus (*Odobenus rosmarus*)—were targeted for closer examination than other species due to their conservation status and importance to Chukchi Sea coastal communities. Nevertheless, substantial analysis effort was also spent measuring Gray whale (*Eschrichtius robustus*), killer whale (*Orcinus orca*), bearded seal (*Erignathus barbatus*), and ringed seal (*Pusa hispida*) acoustic detection distributions. A portion of the data were manually analysed to determine the acoustic occurrence of all marine mammal species and to evaluate and calibrate the automated detectors and classifiers. The automated system was applied to the entire summer datasets to produce indices of acoustic abundance for selected species at each station, and interpolated to create call distribution maps.

The data have characterised migration corridors of bowhead whales in the fall, foraging concentrations of walrus and their in-shore movements due to receding ice, a progressive increase in acoustic detections of more temperate or sub-Arctic species such as fin (*Balaenoptera physalus*), humpback (*Megaptera novaeangliae*), and minke (*Balaenoptera acutorostrata*) whales, and the annual presence of bearded and ringed seals. Fig. 2 shows maps of summer bowhead whale call densities for four consecutive years.

The acoustic recorders have monitored the bowhead whale migration between the Beaufort and Chukchi Seas since 2009. The main corridor for the fall migration, as the whales travel from the Beaufort to the Chukchi, has been observed to generally be north of 71.5° N. A circular array of six recorders (labelled as BG02–BG07 in Fig. 3) is located near the centre of this corridor, and in 2012 two of these recorders (BG04 and BG05) had approximately a third fewer bowhead detections than the adjacent four.
Fig. 2: Summer bowhead whale call densities in 2009, 2010, 2011, and 2012 (by rows from top left). Radial basis-interpolated call densities based on automated and manual call detections at all summer recording stations in the northeastern Chukchi Sea.

Increased vessel traffic to the southeast of this site could possibly have acoustically masked the detections at recorders BG04 and BG05. A generalised linear model was used to investigate the influence of multiple variables on the total call counts at the six recording stations. The analysis showed that sound pressure level (SPL) was the single most influential factor of those considered in the analysis, but the model with SPL and station (i.e. location) was the best fit. When call count versus SPL was compared between the stations, however, a significant difference in call counts was only observed at lower SPLs (Fig. 3).

Fig. 3: Left: Map of 2012 bowhead estimated call counts and summer recorder locations, highlighting the six recorders analysed in the study. Right: Average SPL (100–400 Hz band) vs. bowhead whale call counts for the six recorders of interest.
It appears, therefore, that SPL has a significant effect on the number of detections, as expected, but does not explain the difference observed between the recorders, nor does it suggest acoustic masking. The station variable was then broken down into latitude, longitude, and water depth to examine correlation with call count, SPL, and vessel count. Of those nine correlations, only latitude versus call count was statistically significant (0.83, p-value 0.04). Aerial survey data from other studies also suggests the whales took a more northern route around this array of recorders.

2.2. Vocal repertoire and distribution of bearded seals

Passive acoustic monitoring using multiple recorders has become a feasible method for investigating acoustic behaviour [3] as well as measuring temporal and spatial distributions of marine mammals [1] over large areas and in periods where ship-based or on-ice studies were not previously possible. In a study conducted from 2007 to 2010, male bearded seal vocalisations—detected in year-round underwater recordings at the network of stations described in the previous section—were used to determine the species’ spatial and temporal acoustic distribution at several locations in the northeastern Chukchi Sea. Seasonal patterns in the proportion of different bearded seal call types were also examined, as well as the diel pattern of vocal activity.

Bearded seals were vocally present at all sampled stations; they were identified by their trills, ascents, and moans. Call counts exhibited a consistent seasonal pattern in the study, but overall were highest in the 2007–2008 period. Peaks in calling occurred in spring, coinciding with the bearded seal courting and mating period. Summer had the fewest detected vocalisations. The number of vocalisations increased with the formation of winter pack ice.

Males advertise their breeding condition by producing long underwater trills, notably downsweeping trills longer than 13 seconds [AL2(T)] and trills with ascent/plume [AL1(T)] as illustrated in Fig. 4. The duration of both types of trills increased throughout the winter recording peaking in the mating period (between early May and late June), and then decreasing.

In both summer and winter datasets, a distinct diel pattern was evident in the occurrence of bearded seal vocalisations. Vocalisations occurred most frequently in the early morning, with a peak around 04:00 (AKST), followed by a decrease in the middle of the day, and an increase again at the end of the afternoon. During the mating period, the peak of vocal activity is hypothesised to coincide with times when more females are in the water as previously demonstrated in Svalbard [4].
2.3. Real-time monitoring of walruses

The northeastern Chukchi Sea is a main feeding ground for herds of female and young Pacific walruses during the summer and autumn [6]. Thanks to large-scale PAM programs in the Chukchi Sea [1], their acoustic behaviour is now better understood than before. Data interpretation, however, has been hampered by incomplete characterisation of walrus vocal behaviour and source levels (acoustic pressure at a 1 m reference distance from a point source) in the wild. Source levels are needed to estimate the extent of the walrus communication space and assess how noise from anthropogenic activities can potentially affect group cohesion and behaviour. Source levels of knocking sounds, a dominant call type, have been estimated from wild walruses in the Chukchi Sea [7], but estimates for other call types have been derived from only a single captive adult male [8]. A study was therefore undertaken to obtain a better characterisation of walrus acoustic repertoire and source levels.

In July 2013, two specially developed real-time particle velocity acoustic recording systems (buoys) were used in the northeastern Chukchi Sea to record and localise vocally active walruses in the water near groups hauled out on ice. Each buoy, equipped with a 3-axis dipole sensor and a calibrated omnidirectional hydrophone, was deployed at the water surface and transmitted full acoustic waveform data in real-time to a support skiff (Fig. 5). Successful transmission of the data from the monitoring buoy to the skiff was achieved at ranges up to 1 km.
The range between the recorders, support skiff, and calling animals was usually less than 200 m and typically within tens of metres, which permitted simultaneous visual observations. Five monitoring sessions yielded 14 hours of recordings. Calling walruses in water were localised in two dimensions in near real-time using cross-fixes of acoustic bearings from the dipole sensors. Source levels were estimated by adding modelled frequency-dependent transmission losses to the received levels in each 1/3-octave-band obtained from the calibrated omnidirectional hydrophone.

Focal groups were largely biased in favour of males. Knock and bell calls were the dominant vocalisations, possibly because of the skewed sex ratio. A variety of grunt/barks was also recorded. All recordings were obtained from socially active walrus around haul-outs, not from feeding or travelling individuals. The source levels of measured knocks were consistent with another study in the Chukchi Sea conducted using bottom-mounted recorders [7]. Grunts had the lowest source levels; they are presumably used for short-range communication and are most at risk from masking by anthropogenic noise and associated reduction in communication space. Our measurements provide the first reported source levels of walrus grunts in the wild.

The real-time monitoring buoys enabled concurrent acoustic and visual observations. The buoys were small enough to fit in an inflatable boat and light enough to deploy by hand and relocate easily. Walruses were unresponsive to the presence of the buoys, which makes this equipment well suited to monitoring animals in a social group without disrupting them appreciably. The system was less adequate for tracking travelling and foraging walrus that rapidly fell out of visual/acoustic range. Localisation required calls with high signal to noise ratios.

3. SUMMARY

The three case studies presented give a sense of the power of acoustic monitoring of marine mammals as a tool for long-term characterisation of their distribution and behavioural patterns or detailed study of their vocalisations. Readily available automated detection and classification procedures enable the processing of large volumes of data, which, combined with advances in recorder autonomy and resilience, makes winter monitoring studies in Arctic waters feasible and practical. The level of effort required by large-scale multi-year acoustic monitoring programs is considerable, but the advancement
in knowledge of the marine environment achieved through these efforts is unprecedented. On the other hand, much simpler monitoring installations can answer key bio-acoustical questions at the individual or small group scale in the wild. Passive acoustic monitoring technology has matured into a reliable, practical, and valuable tool to understand and help conserve marine mammals.

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PASSIVE ACOUSTIC SOURCE LOCALISATION METHODS FOR THE NONINTRUSIVE MONITORING OF ECHOLOCATING DOLPHINS IN THE WILD

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Abstract: Dolphins probe the underwater environment using sequences of short-duration underwater sound pulses (or clicks). Passive sonar source localisation algorithms, which are based on measuring the range differences of a click at pairs of sensors in an acoustic receiving array, are shown to provide estimates of the instantaneous source positions of all clicks in a sequence, even when three dolphins are echolocating at the same time. The simplest (wide aperture) array configuration for passive sonar source localisation consists of three sensors widely spaced along a straight line. The conventional localisation method of passive ranging by wavefront curvature provides estimates of the instantaneous range (with respect to the middle sensor) and relative bearing (with respect to the array axis) of the source. However, the range estimates, when calculated using the conventional method, are found to have significant bias due to a small lateral displacement of the middle sensor away from the array axis. A modified method, which overcomes the range bias problem when the known sensor positions are not collinear, produces identical values for the source positions when calculated using two other methods for passive sonar source localisation by range differences, namely the line of position and circle intersection methods. Source localisation is then extended from two- to three-dimensions by applying a single sensor multipath time delay method to estimate the source altitude. This method requires measuring the time delay between the click arriving by a sea surface reflected propagation path and that arriving by the direct travel path from source to sensor. With this method, the altitude (or corresponding depth) of an echolocating dolphin is determined to within ±5 cm, even for closely-spaced echolocating dolphins in a pod at ranges in excess of 275 m. Finally, the nonintrusive passive acoustic source localisation methods presented here constitute a powerful scientific toolset for studying the behaviour of echolocating dolphins in the wild.

Keywords: Dolphin sonar, passive sonar source localisation, wide aperture array
1. INTRODUCTION

Dolphins explore their aquatic habitats using short-duration underwater sound pulses (or clicks) for echolocation that enable them to navigate and avoid collisions with natural objects, and to detect, localise and discriminate between prey, predators, and companions, even while swimming at night or in turbid water. Traditionally, researchers apply passive sonar signal analysis methods to underwater acoustic data recorded when dolphins are present in order to study the properties of dolphin biosonars and to characterise their signal emissions [1]. These methods work best when only one dolphin is echolocating at any one time. Problems experienced by marine scientists [1, 2] when analysing echolocation clicks from recordings of free-ranging dolphins are:

(1) it is generally unknown which dolphin is producing the recorded clicks and how many animals are echolocating;

(2) the peak-to-peak source level of clicks cannot be estimated with accuracy because the distance of the dolphin from the hydrophone is unknown; and

(3) the direct path and multipath arrivals are not resolvable when the source and receiver are close (within 1 m) of the sea surface.

These problems are also encountered in the present experiment. For example, Fig. 1 shows multiple regular click sequences observed during a short (1 s) time interval. It is uncertain how many dolphins are present and their respective distances from the hydrophone. As well as the direct acoustic propagation path of the click signals from the underwater sound projector (dolphin active sonar) to the hydrophone receiver, other multipath (indirect) arrivals are also present due to some of the signal propagation paths involving reflections from the sea surface and/or the sea floor.

Fig. 1. Variation with time of the output of a hydrophone when numerous dolphins are echolocating at the same time. The observation time interval is 0.94 s.
In this paper, these problems are solved by processing the data from a collinear array of three widely-spaced hydrophones configured as a wide aperture array. The source position of each click signal arriving at the array is estimated using passive sonar methods which require range-difference measurements between adjacent pairs of hydrophones. In Section 2, these two-dimensional source localisation methods are reviewed and then applied to the localisation of seven echolocating dolphins in the wild during a 30 s observation time interval. In Section 3, the localisation of echolocating dolphins is extended from two to three dimensions by using a single hydrophone multipath passive ranging method to estimate the source altitude (with respect to the sea floor). Section 4 provides the Conclusions.

2. TWO DIMENSIONAL PASSIVE SONAR SOURCE LOCALISATION METHODS BASED ON RANGE DIFFERENCE MEASUREMENTS

The signals emitted by a source arrive at the sensors of a receiving array at times dependent on the source-sensor geometry and the characteristics of the propagation medium. Differences in arrival times of the source signals at pairs of sensors are measurable and so the location of the source can be estimated if the sensor positions are known. In an isospeed signal propagation medium, the time difference of arrival (TDOA) at a pair of spatially-separated sensors is proportional to the difference in the two source-sensor ranges, termed the range difference (RD). The range difference is equal to the product of the signal’s constant speed of travel and the time difference of arrival. When the source and sensors are in the same plane, the locus of possible source positions for a constant RD is a hyperbola, and the location of the source can be estimated as an intersection of the set of hyperbolas defined by the RD measurements of various sensor pairs [3].

The passive sonar problem of source localisation using range-difference measurements is solved here for the two dimensional case. Let the source be at unknown position \((x_s, y_s)\) and the sensors be at known positions \((x_i, y_i)\), where \(i = 1, 2, 3\) - see Fig. 2.

![Diagram of source-sensor geometry](image)

*Fig. 2. Source-sensor geometry for source localisation using range-difference measurements.*
The distances from the origin to the sensors are $R_{10}$, $R_{20}$ and $R_{30}$, and the distance to the source is $R_{s0}$. The distance between the $i$th sensor and the source is $R_{si}$. Let $r_{ij}$ be the RD for sensors $i$ and $j$ where

$$r_{ij} = R_{is} - R_{js} = \left( (x_i - x_s)^2 + (y_i - y_s)^2 \right)^{\frac{1}{2}} - \left( (x_j - x_s)^2 + (y_j - y_s)^2 \right)^{\frac{1}{2}}. \quad (1)$$

For $i, j = 1, 2, 3$

$$\left( R_{is} \right)^2 = \left( r_{ij} + R_{js} \right)^2 = r_{ij}^2 + 2r_{ij}R_{js} + R_{js}^2. \quad (2)$$

Also,

$$R_{is}^2 = (x_i - x_s)^2 + (y_i - y_s)^2 = x_i^2 - 2x_ix_s + x_s^2 + y_i^2 - 2y_iy_s + y_s^2, \quad (3)$$

$$\therefore R_{is}^2 = R_{so}^2 + R_{so}^2 - 2x_ix_s - 2y_iy_s, \quad (4)$$

Equating (2) and (4) gives

$$2x_ix_s + 2y_iy_s = R_{so}^2 + R_{so}^2 - r_{ij}^2 - 2r_{ij}R_{js} - R_{js}^2. \quad (5)$$

Setting $i = j$ and noting $r_{jj} = 0$,

$$2x_ix_s + 2y_iy_s = R_{jo}^2 + R_{so}^2 - R_{js}^2. \quad (6)$$

Subtracting (6) from (5),

$$2(x_i - x_s)x_s + 2(y_i - y_s)y_s = R_{so}^2 - R_{jo}^2 - r_{ij}^2 - 2r_{ij}R_{js}. \quad (7)$$

For the present case, $j = 1,$ and $i = 2, 3$. Hence, the fundamental equations for source localisation from range differences are given by:

$$\left(x_2 - x_1\right)x_s + \left(y_2 - y_1\right)y_s = \frac{1}{2}\left(R_{20}^2 - R_{10}^2 - r_{12}^2\right) - R_{1s}r_{2s}, \quad (8)$$

$$\left(x_3 - x_1\right)x_s + \left(y_3 - y_1\right)y_s = \frac{1}{2}\left(R_{30}^2 - R_{10}^2 - r_{13}^2\right) - R_{1s}r_{3s}. \quad (9)$$

Different source localisation algorithms that are based on range difference measurements between pairs of sensors represent different strategies for solving the fundamental equations for source localisation from range differences which are given by (8) and (9). In other words, the line of position, circle intersection and passive ranging by wavefront curvature methods differ only in their approaches to solving these fundamental equations. For instance, when the source and sensors are in the same plane, the source location coincides with the point of intersection of three circles, which can be represented mathematically as:

$$(x_i - x_s)^2 + (y_i - y_s)^2 = R_{so}^2, \quad i = 1, 2, 3. \quad (10)$$

The circle intersection method can be derived from (10), or by rearranging the fundamental equations (8), (9) for source localisation from range differences, viz

$$R_{20}^2 - r_{21}^2 - R_{10}^2 - 2R_{1s}r_{2s} + 2x_ix_s + 2y_1y_s = 2x_2x_s + 2y_2y_s, \quad (11)$$

$$R_{30}^2 - r_{31}^2 - R_{10}^2 - 2R_{1s}r_{3s} + 2x_ix_s + 2y_1y_s = 2x_3x_s + 2y_3y_s. \quad (12)$$

A new source localisation method, termed the modified method for passive ranging by wavefront curvature, has been formulated for arbitrary (but known) sensor positions [4]. The source position estimates are found to be equivalent to those calculated using the line of position and circle intersection methods [5]. The conventional method for passive ranging by wavefront curvature is a particular case of the modified method when the sensor positions are collinear. For the general problem of passive localisation of an acoustic source where the application is three dimensional by nature, the line of position method has been shown mathematically to be equivalent to the spherical interpolation method [6].

3. TWO DIMENSIONAL LOCALISATION OF ECHOLOCATING DOLPHINS
The time of arrival of a click at each sensor in the hydrophone array is recorded in a database which covers a 30 s observation time interval. The TDOA data for the clicks are then used to estimate the acoustic source positions of echolocating dolphins. Figure 3 shows the results of applying the line of position method (or, equivalently, the modified method for passive ranging by wavefront curvature, or else the circle intersection method) to localise echolocating dolphins. The positions of the sensors are shown as black filled circles, with the array axis inclined at an angle of 28.5° with respect to the x axis (or, 61.5° when measured clockwise with respect to True North). The Cartesian coordinates of the instantaneous source positions are colour coded for each of seven dolphins. (Note that in Fig. 1, there are three echolocating dolphins: \textit{Alfa}, \textit{Bravo}, and \textit{Echo}.) Dolphin \textit{Alfa} is at a range of 318 m from the origin (i.e. the position of the middle sensor) with a bearing of 146° when measured anticlockwise with respect to the direction of the array axis. Similarly, dolphin \textit{Echo} is at a range of 33 m and bearing 322°, with \textit{Bravo} at 91 m and bearing 315°. In the case of \textit{Echo} (short range, 33 m) and \textit{Bravo} (medium range, 91 m), the instantaneous positions change with time because the dolphins are moving. For \textit{Alfa} (long range, 318 m), the movement of the dolphin can be observed in bearing but not in range because the range resolution is too coarse. The observed variation with range of the source range error (after the range estimates are detrended) agrees with theoretical prediction [7]: a 0.025 m range error for a dolphin at close range (33 m), increasing to 0.19 m at medium range (91 m), and 2.6 m at long range (318 m).

Fig. 3. Cartesian coordinates (in metres) of the source positions of 640 dolphin clicks.

4. THREE DIMENSIONAL LOCALISATION OF ECHOLOCATING DOLPHINS

Given a flat sea surface, the multipath time delay \( \tau_{s,d} \) of the arrival of the surface-reflected multipath relative to the direct path is given by

\[
\tau_{s,d} = \frac{R_s - R_d}{c},
\]  

(13)
where $R_s$ and $R_d$ are the ray path lengths of the surface-reflected multipath and the direct propagation path, respectively. Using Pythagorean triangle geometry, it can be shown that

$$R_s = \sqrt{R_d^2 + 4(h_w - h_r)(h_w - h_r)},$$

(14)

where $h_w$, $h_r$, $h_r$ are the altitudes (measured from a flat sea floor) of the sea surface, transmitter, and receiver, respectively. Note that $h_w$ is equivalent to the water depth. It can be shown, by combining (13) and (14) and rearranging so $h_r$ is the subject, that

$$h_r = \frac{4h_w(h_w - h_r) + R_d^2 - (c\tau_{s,d} + R_d)^2}{4(h_w - h_r)}.$$

(15)

Now $\hat{h}_r$ can be estimated from (15), given $h_w = 20$ m, $h_r = 1$ m, $c = 1521$ m/s, with $\hat{R}_d$ estimated using a wide aperture array modified passive sonar ranging method, and $\hat{\tau}_{s,d}$ measured by taking the difference in the signal arrival times of the surface-reflected multipath and the direct path. Figures 4 and 5 show the waveforms of a click, which is initially emitted by echolocating dolphin Romeo and subsequently received at the middle hydrophone, after propagating along the direct ray path (Fig. 4), or along the indirect surface-reflected ray path (Fig. 5). Note that the phase of the incident pulse undergoes a change of $\pi$ radians when reflected from the sea surface. In Fig. 4, the time of arrival of the minimum of the first trough is at 10min:22.3184151sec. In Fig. 5, the time of arrival of the first peak is 10min:22.3190976sec. Also, other pulses, which are attributed to out-of-plane surface reflections, arrive immediately after the in-plane surface reflected multipath signal. For this click, $\hat{\tau}_{s,d} = 0.6825$ ms and $\hat{R}_d = 270.8$ m, so the altitude estimate of Romeo is $\hat{h}_r = 12.59$ m.

The passive sonar methods considered here represent a high resolution source localisation toolset for the nonintrusive scientific study of the behaviour of echolocating dolphins. For example, these techniques enabled two individual echolocating dolphins within a pod (Romeo and Sierra) to be resolved in space (range, bearing and depth) even at long ranges -

Fig. 4. Click waveform – Direct path

Fig. 5. Click waveform – Multipath arrival
see Figs 6 (resolved in altitude) and Fig. 7 (resolved in planar position). Also, these methods allow insights into how dolphins might use their echolocation capability when in a pod. An example of cooperative behaviour is observed for the two pod dolphins, *Romeo* (range 268 m, bearing 142°, altitude 12.7 m) and *Sierra* (285 m, 143°, 13.1 m) which have closely matched click waveforms, so *Romeo* stops projecting its biosonar pulse transmissions when *Sierra* starts echolocating with the overlap period of the two sequences being one sixth of a second – see Fig. 6. This behaviour suggests that, at any one time, only one dolphin in the pod echolocates using the monostatic sonar principle. Additionally, during this time, the other dolphins might conceivably operate their biosonars in passive mode. In other words, echolocation by a pod of dolphins is based on the bistatic sonar principle, i.e. one active source (acoustic projector) and numerous spatially distributed receivers.

Fig. 6. Altitude estimates vs click number of signals emitted by dolphins Romeo and Sierra.

Fig. 7. Cartesian coordinates (in metres) of signals emitted by dolphins Romeo and Sierra.
5. CONCLUSIONS

The positions of echolocating dolphins in the wild can be estimated using passive sonar source localisation methods. As predicted by theory, the source range errors are observed to depend on the square of the range of the source, whereas the source bearing errors are range independent. Estimates of the source positions in two dimensional space can be extended to three dimensional space by estimating the instantaneous source altitude (to within ±5 cm), using a single sensor multipath time delay method.

The precision of the source position estimates of echolocating dolphins using the passive sonar source localisation methods presented here is unprecedented. These passive sonar methods represent a high resolution passive sonar source localisation toolset for scientifically studying the behaviour of individual dolphins within a pod, even at long ranges.

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REFERENCES

PASSIVE ACOUSTIC MONITORING OF MARINE MAMMALS USING A TERNARY ARRAY

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\textbf{Abstract:} There is a growing interest in the monitoring of vocalising marine mammals. Their behaviour is studied in the context of climatic changes and human civil and military activities. Passive acoustic surveying methods are preferred since they do not affect animals’ natural behaviour. For this reason, a variety of passive methods have been proposed but most are expensive and involve complicated logistics. In this paper, a passive, simple and flexible ternary array is proposed for tracking marine mammals from a small vessel or an anchored platform or floating platform. The system consists of three surface buoys, equipped with array’s hydrophones, pressure sensors and global positioning system (GPS) receivers. One hydrophone is a vector hydrophone. The floating buoys are deployed from the boat, connected in chain with a rope, and float with the currents and winds. The array can stay in a position for a continuous and prolonged length of time without the necessity of human operators. The received data are post-processed to track single or multiple marine mammals. A simpler version of the system utilizes an electrical cable connecting all buoys with the platform. The cable can be used to supply power to each buoy and transmit data from them. It is envisioned that the array of approximately 40m length will allow sufficient accuracy of target range estimation, while target angular positions are obtained using the vector hydrophone. The simulation results support that expectation.

\textbf{Keywords:} Marine Mammal Monitoring, Passive Localization, Flexible Ternary Array, Acoustic Vector Sensor
1. INTRODUCTION

The research on monitoring of marine mammals includes species diversification monitoring, environment analysis and biological evolution [1, 2]. Monitoring methods and diverse data-loggers have been developed rapidly over the last several decades [3], and a variety of marine mammals’ sounds were recorded and classified in different parts of the world [4]. However, most of those methods are expensive and involve complicated logistics, which limit their wide applications. In this paper, a passive, simple and flexible ternary array is proposed for tracking marine mammals from a small vessel or an anchored or floating platform. It is envisioned that the array of approximately 40m length will enable the achievement of sufficient accuracy of target position tracking. Target angular positions (elevation and azimuth angles) are obtained using the vector hydrophone, while range can be calculated based on relative signal delays between hydrophones. The simulation results support the feasibility of the proposed system.

2. MARINE MAMMAL SIGNALS

Marine mammals’ signals can be classified into whistles and clicks. Both of them play an important role in their survival [5]. Whistles are narrowband frequency modulated signals with spectrum within an audio range. Typically, whistles are generated in sequence of 1 to 9, each whistle of certain duration and characteristics. Duration of a whistle is approximately several tenths of a second with interval between whistles of tenths of a second to 2 seconds. Whistle signals are designed for long range and broad-beam broadcast that requires high power transmission signals. The linear frequency modulation (LFM) is often used to model marine mammals’ whistle signal $s(t)$, as given by Eq.1.

$$s(t) = A \exp \left[ j \left( 2\pi f_0 t + \pi \beta t^2 \right) \right] \quad t \in [-\tau/2, \tau/2] \quad \text{and} \quad 0 \quad \text{other} \quad (1)$$

where $A$ is the amplitude of the signal, $j = \sqrt{-1}$, $\beta = B/\tau$ is the slope of frequency modulation, $B = f_H - f_L$ is the signal bandwidth with the frequency range from $f_L$ to $f_H$, $f_0$ is the central frequency, and $\tau$ is the duration of the signal.

The click is a short pulse of several hundredths of millisecond duration. The clicks are generated in series of duration between 100 and 200 milliseconds and occupying a frequency up to 140 kHz. They are used to localize prey by echolocation. They have high intensity and propagate over narrow-beam.

3. METHOD

The passive monitoring is based on signals generated by marine mammals acting as sound sources or monitored targets.

3.1 Passive Acoustic Monitoring of Marine Mammals Using a Ternary Array

The schematic diagram of the proposed monitoring system is shown in Fig.1. The system consists of three surface buoys, data recorders, global positioning system (GPS) receivers, and pressure sensors. Buoys “A” and “B” are equipped with hydrophones, whereas the reference buoy “R” has an acoustic vector hydrophone (AVH). AVH [6]
measures the water particle velocity vector $V$ and scalar pressure $p$. The velocity vector $V$ allows source (target) angular localization in terms of elevation and azimuth angles. The received data are processed to track single or multiple marine mammals. The array can stay in position for a continuous and prolonged length of time without the necessity of human operators.

**Fig. 1: Schematic diagram of passive acoustic monitoring of marine mammals using a flexible ternary array**

### 3.2 Direction of arrival (DOA) Estimation

Particle velocity vector $V$ has 3 orthogonal components, that is

$$\begin{cases}
  v_x = v(t) \cos \theta \cos \alpha \\
  v_y = v(t) \sin \theta \cos \alpha \\
  v_z = v(t) \sin \alpha
\end{cases} \tag{2}$$

where $v(t) = |V| = \sqrt{v_x^2 + v_y^2 + v_z^2}$ is the speed of particle, $\theta$ is the azimuth and $\alpha$ is the elevation angle of a target. The particle speed $v(t)$ is proportional to pressure $p(t)$ as given by Eq.3:

$$v(t) = \frac{1}{z} p(t) \tag{3}$$

where $z = \rho c$ is acoustic impedance of water, $\rho$ is the density of water, and $c$ is sound propagation speed in water.

Considering the noise at the receiver, the outputs of AVH can be expressed as follows:

$$\begin{cases}
  p(t) = x(t) + n_p(t) \\
  v_x(t) = \frac{1}{z} x(t) \cos \theta \cos \alpha + n_x(t) \\
  v_y(t) = \frac{1}{z} x(t) \sin \theta \cos \alpha + n_y(t) \\
  v_z(t) = \frac{1}{z} x(t) \sin \alpha + n_z(t)
\end{cases} \tag{4}$$
where $x(t)$ is the target sound pressure at the receiver; under realistic conditions noises are present $\mathbf{V}$ components in Eq.2, that is $n_x(t)$, $n_y(t)$ and $n_z(t)$ as well as $n_p(t)$ shown in Eq.4.

Instantaneous acoustic sound intensity (scalar) is

$$I(t) = p(t)v(t)$$  \hspace{1cm} (5)

One can introduce “superficial” sound intensity components $I_x$, $I_y$ and $I_z$ which are aligned with velocity vector components, such that actual sound intensity as given in Eq.5 is

$$I = \sqrt{I_x^2 + I_y^2 + I_z^2}$$

Time-averaged acoustic intensity (scalar) is

$$\bar{I}(t) = \frac{p(t)v(t)}{2}$$  \hspace{1cm} (6)

Expanding Eq.6

$$\bar{I}_x = \frac{p(t)v_x(t)}{2} = \frac{1}{z} x^2(t) \cos \theta \cos \alpha + n_p(t)n_x(t) + \frac{1}{z} n_p(t)x(t) \cos \theta \cos \alpha + n_x(t)x(t)$$  \hspace{1cm} (7)

$$\bar{I}_y = \frac{p(t)v_y(t)}{2} = \frac{1}{z} x^2(t) \sin \theta \cos \alpha + n_p(t)n_y(t) + \frac{1}{z} n_p(t)x(t) \sin \theta \cos \alpha + n_y(t)x(t)$$  \hspace{1cm} (8)

$$\bar{I}_z = \frac{p(t)v_z(t)}{2} = \frac{1}{z} x^2(t) \sin \alpha + n_p(t)n_z(t) + \frac{1}{z} n_p(t)x(t) \sin \alpha + n_z(t)x(t)$$  \hspace{1cm} (9)

It is assumed that $n_p(t)$, $n_x(t)$, $n_y(t)$, $n_z(t)$ and $x(t)$ are weekly dependent variables, therefore, the first terms in Eq.7, Eq.8, and Eq.9 are much greater than other terms particularly for the high signal-to-noise ratio (SNR), and the three equations can be simplified as

$$\bar{I}_x = \frac{1}{z} x^2(t) \cos \theta \cos \alpha + \Delta_x$$  \hspace{1cm} (10)

$$\bar{I}_y = \frac{1}{z} x^2(t) \sin \theta \cos \alpha + \Delta_y$$  \hspace{1cm} (11)

$$\bar{I}_z = \frac{1}{z} x^2(t) \sin \alpha + \Delta_z$$  \hspace{1cm} (12)

where $\Delta_x$, $\Delta_y$, $\Delta_z$ are additive errors.

The estimated elevation angle $\hat{\alpha}$ and estimated azimuth angle $\hat{\theta}$ can be obtained

$$\hat{\alpha} = \arcsin \left( \frac{\bar{I}_z}{\bar{I}} \right) = \arcsin \left( \frac{zp(t)v_z(t)}{p^2(t)} \right)$$  \hspace{1cm} (13)

$$\hat{\theta} = \arctan \left( \frac{\bar{I}_y}{\bar{I}_x} \right) = \arctan \left( \frac{I(t)v_y(t)}{I_x(t)v_x(t)} \right)$$  \hspace{1cm} (14)

In the presence of several targets transmitting at the same time, the reading of AVH is the combination of individual targets’ angular position vectors and does not represent any specific target. Therefore, time-averaged acoustic intensity is not suitable for such a situation. However, complex-averaged acoustic intensity [6] is proposed to obtain the angular position at different frequency, which provides the possibility to identify multi targets overlapping in time domain not necessary in frequency domain. The schematic diagram of calculation of complex-averaged acoustic intensity is shown in Fig.2.
$P(f)$, $V_i(f)$ are the frequency spectra of $p(t)$ and $v_i(t) (i=x,y,z)$, and the complex cross spectrum between sound pressure and particle velocity are

$$S_{pvi}(f) = P(f)V_i^*(f) (i=x,y,z) \quad (15)$$

where “*” indicates complex conjugate.

Eq.16 shows Average over frequency with rectangular window

$$\langle S_{pvi} (f) \rangle = \langle P(f)V_i^*(f) \rangle (i=x,y,z) \quad (16)$$

The outputs of calculation of complex-averaged acoustic intensity are

$$I_{Rx} (f) = \text{Re} \{ \langle P(f)V_x^*(f) \rangle \} \quad (17)$$

$$I_{Ry} (f) = \text{Re} \{ \langle P(f)V_y^*(f) \rangle \} \quad (18)$$

$$I_{Rz} (f) = \text{Re} \{ \langle P(f)V_z^*(f) \rangle \} \quad (19)$$

$$I_{Rp} (f) = \text{Re} \{ \langle P(f)P^*(f) \rangle \} \quad (20)$$

It follows from Eq.3 that in frequency domain the spectra of $P(f)$ and $V_i(f)$ are identical except for a scaling coefficient ($1/z$). It can be known from cross spectrum analysis that the co-spectrum (the real part) is the in-phase signal and the other spectrum (complex) is the out-of-phase signal.

The elevation and azimuth angles of the target can be estimated using Eq.17 to 20

$$\hat{\alpha} (f) = \text{arcsin} \left( \frac{I_{Rx}}{I_{Rp}} \right) = \text{arcsin} \left( \frac{\text{Re} \{ z \ast \langle P(f)V_x^*(f) \rangle \}}{\text{Re} \{ \langle P(f)P^*(f) \rangle \}} \right) \quad (21)$$

$$\hat{\theta} (f) = \text{arctan} \left( \frac{I_{Ry}}{I_{Rx}} \right) = \text{arctan} \left( \frac{\text{Re} \{ \langle P(f)V_y^*(f) \rangle \}}{\text{Re} \{ \langle P(f)V_z^*(f) \rangle \}} \right) \quad (22)$$

Let’s assume that signals from targets with different spectra “signatures” are received with a time domain observation window of $T_{ob}$ for which the targets can be considered as stationary. The aggregated “target” angular position can be obtained at different frequency based on Eq. 21 and Eq. 22. The number of occurrences of the same angular (elevation or azimuth) position is counted to construct a histogram for bar-graph presentation. The persistent occurrence at a certain angle will be considered as an indication of a specific target. This approach is applicable for several targets with distinct spectra signatures. For overlapping spectra signatures the targets are recognized based on the relative spectra levels.

### 3.3 Passive ranging

The general geometry of array and simulated target (whale) is shown in Fig.3.
The position of vector hydrophone AVH \((H_0)\) is set as the origin \((0,0,0)\), and the position of the two hydrophones are known \(H_1(x_1,y_1,z_1)\) and \(H_2(x_2,y_2,z_2)\) as given by GPS and pressure sensor readings. The position of the target can be written as \(T(x,y,z)\). The vector hydrophone will provide the estimated elevation and azimuth angles, named as \(\hat{\alpha}\) and \(\hat{\theta}\). The target signal is received by 3 hydrophones with AVH \((H_0)\) considered the reference. The delay differences \(\Delta T\) between 3 hydrophones are measured using cross correlated method and are related to the target position as given by Eq.23.

\[
\begin{align*}
\tan \hat{\theta} &= \frac{y}{x} \\
\tan \hat{\alpha} &= \frac{z}{\sqrt{x^2 + y^2}} \\
\Delta T_{i,o} &= \left(\sqrt{(x-x_i)^2 + (y-y_i)^2 + (z-z_i)^2} - \sqrt{(x-x_o)^2 + (y-y_o)^2 + (z-z_o)^2}\right) / c \ (i = 1 \text{ or } 2)
\end{align*}
\]

where \(c\) is sound propagation speed in water; \(\Delta T_{1,o}\) or \(\Delta T_{2,o}\) are corresponding delays; each of them can be used for range calculation. There is also a possibility of using both for improvement estimation.

Solving the set of equations for \(x\), \(y\) and \(z\), the target range \(r\) is

\[
r = \sqrt{x^2 + y^2 + z^2}
\]

### 4. NUMERICAL SIMULATIONS

In the following simulations, two large odontocetes generating whistles, modelled by Eq.1, are tracked by the proposed method. The simulation conditions are indicated in Table 1. No ambient noise and multi-path interference were considered. It is assumed that
the trajectory of each target has three stages: diving vertically into deep water, moving horizontally at a certain depth, and ascending to the sea surface; all stages with constant speed. The received targets signals are observed within the observation window of $T_{ob}$=1s, in which the targets are considered to be stationary. The worst condition assumed is that the shorter duration signal from target #1 with $\tau_1=0.4$s is within the longer duration signal of $\tau_2=0.8$s from target #2. The exact time position of such overlapping signals within the observation window doesn’t affect the computation results. This procedure is repeated several times for signals generated by targets with variable spatial positions. The results are shown in Fig.4 for 40 sequential observation windows.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Targets</th>
<th>Frequency range of the signal (kHz) $[f_L, f_H]$</th>
<th>Range of elevation angle (degree) $\alpha$</th>
<th>Range of azimuth angle (degree) $\theta$</th>
<th>Duration of signal (s) $\tau$</th>
<th>SIR (Signal to interference ratio) (dB) $10\log_{10}(P_2/P_1)$</th>
<th>Span for moving average</th>
</tr>
</thead>
<tbody>
<tr>
<td>#1</td>
<td>[4,12]</td>
<td>[30,67]</td>
<td>[30,34]</td>
<td>0.4</td>
<td></td>
<td>10</td>
<td>5</td>
</tr>
<tr>
<td>#2</td>
<td>[5,8]</td>
<td>[50,87]</td>
<td>[50,54]</td>
<td>0.8</td>
<td></td>
<td>10</td>
<td>5</td>
</tr>
</tbody>
</table>

**Table 1: Parameters of the simulation**

![Graph](image1)

**Fig. 4: Simulation results for 2 targets: (a) variable elevation (b) variable azimuth**
In Fig.4, the “circle” shows the assumed angular positions of each target, and the “dot” indicates the estimated positions. It can be seen that all the positions of target 1 have been estimated accurately, whereas there are several inaccurate points for target 2. This is because by targets characteristics as given in Table 1, that is spectral signatures overlap and signal-to-interference ratio (SIR).

5. CONCLUSION

This paper devises a scenario for passive marine mammal monitoring with flexible ternary array. The extensive simulation has shown that a relatively simple and flexible passive monitoring system can be successfully used for marine vocalising mammal monitoring. Future developments will include effects of ambient noise, multi-path and tracing of multitude of targets.

ACKNOWLEDGEMENT

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REFERENCE

USING ACOUSTICS TO REVEAL POPULATION STRUCTURE OF THE ELUSIVE BLUE WHALE.

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Abstract: Our inability to directly observe animals in marine environments has limited our understanding of elusive species. Passive acoustic monitoring is a cost-effective method to passively monitor vocal marine mammals over large spatial and temporal scales, and has proven a powerful tool in revealing distribution patterns, in particular for migratory species. The blue whale, although the largest animal alive, can exhibit elusive behaviour. Their pelagic habitat, wide dispersal and low population densities make field observations difficult. The pygmy blue whale (Balaenoptera musculus brevicauda), a subspecies, is listed as data deficient and occurs in the southeast Indian Ocean, yet little is known about their occurrence in the southwest Pacific Ocean. Pygmy blue whales produce regionally-specific calls including the Madagascan, Sri Lankan, Australian, New Zealand and Solomon type calls. Passive acoustics data was recorded at five sites (two in the southeast Indian Ocean and three in the southwest Pacific Ocean) between February and August (2009 and 2010). This data will be analysed using automated methods to detect the occurrence of pygmy blue whales off the Tasman Sea and furthermore outline the population structure across the south Indian and Pacific Ocean basins. This paper explains the methods that will be used in this study.

Keywords: PAM, marine mammal, pygmy blue whale, Australia, New Zealand, conservation tool.
1. INTRODUCTION

Passive acoustic monitoring (PAM) is a cost-effective method to passively monitor vocal marine mammals over large spatial and temporal scales. It is not limited by direct field access to animals, is not weather or daylight dependant, and has proven a powerful tool in revealing distribution and movement patterns, in particular for highly migratory marine mammals. The blue whale is the largest animal on earth, yet their elusive behaviour, pelagic habitat, extensive migrations and low population densities, due to its exploitation during the 20th century, make field studies difficult. In the past, knowledge about their distribution was derived from whaling records and targeted aggregation areas, but in recent years PAM has been used to outline spatial and temporal occurrence and infer migration patterns and habitat use.

For baleen whales, acoustic behaviour plays an important role in communication and in particular in courtship behaviour. Baleen whales tend to have acoustic repertoires that are generally stereotypical and repetitive and can be used for species identification. There are two blue whale subspecies found in the Southern Hemisphere, the pygmy blue whale (*Balaenoptera musculus brevicauda*) and the Antarctic or true blue whale (*B.m. Intermedia*), classified under the IUCN Red List of Threatened Species, as data deficient and critically endangered respectively. Unlike the Antarctic blue whale, pre-exploitation population numbers of the pygmy blue whale are not known and little information is available on abundance and distribution. In recent years separate populations of pygmy blue whales have been identified through PAM, based on geographic variation in call characteristics [1-3].

There are five known pygmy blue whale ‘acoustic populations’ associated with geographic regions in the Southern Hemisphere. Within the Indian Ocean pygmy blue whales produce three different types of calls: the Sri Lankan, Madagascan [2] and Australian call type [4]. Generally pygmy blue whales are found in low latitudes during the austral winter where they are thought to breed for example off Indonesia [5]. During the austral summer they are aggregating in potential feeding grounds in higher latitudes including areas off Crozet Islands [6], the Madagascan Basin [3], the Perth Canyon [7], and the Bass Strait [8].

Less is known about the acoustic population structure of pygmy blue whales in the SW (?) Pacific Ocean; to date, two types of calls are known, the Solomon [9] and New Zealand [10-12] call types. Little information is known about distribution patterns and breeding sites for these acoustics populations, and only recently a feeding area was described off western New Zealand [13]. In this study we aim to identify whether pygmy blue whales are present within the Tasman Sea, and as call types are based on proximity to geographical location, and hope to clarify whether these whales belong to the Australian or New Zealand acoustic populations.

2. METHODS

2.1. Data

To detect the presence of pygmy blue whales, passive acoustic data were recorded between February and August in 2009 or 2010 at five locations: two in the southeast
Indian Ocean (Perth Canyon, Western Australia and Bass Strait, Victoria), and three locations in the southwest Pacific Ocean (the Tasman Sea, Eastern Australia, off Tonga and off Samoa; Figure 1). At three of the five locations (Perth Canyon, Bass Strait and Tasman Sea) single fixed hydrophones, a part of the Australian Integrated Marine Observer System (IMOS), were used between February and August 2010. The IMOS hydrophones recorded ocean sounds for 500s every 900s at a sampling rate of 6,000 Hz. At the two remaining locations, Tonga and Samoa, single fixed hydrophones - deployed by Oregon State University (OSU) - were used. The OSU hydrophones recorded continuous ocean sound at a sampling rate of 250 Hz off Tonga and 1,000 Hz off Samoa. Acoustic data were collected off Tonga between February and August 2009 and off Samoa between February and August 2010.

2.2 Calls

An automated spectrogram correlation method will be used to detect specific pygmy blue whale call types in Ishmael (V.1.0) [14]. All data will be resampled to 250 Hz before analysis. A detector template for each call type (Australia and New Zealand) will be created based on time and frequency characteristics unique to each call type. The detector created for the Australian pygmy blue whale call will target the third harmonic of this call with energy between 65 and 71 Hz. The call is 15 s in duration with a bandwidth of 3 Hz. The New Zealand pygmy blue call consists of two parts and both parts will be used for the detector. The first part of the call, between 23-26 Hz, is approximately 15s in duration while the second part of the call, a harmonic at 17-20 Hz, is approximately 20s long. A 3 Hz bandwidth will be used for part 1 and 2 of the call. To eliminate continuous interfering tones like ships, equalization time constants of 40 s and 30 s will be used for the Australian and New Zealand pygmy blue whales calls, respectively.

Both Australian and New Zealand pygmy blue whale call detectors will be run across all five sites (Perth Canyon, Bass Strait, Tasman Sea, Tonga and Samoa). All calls that are detected will be checked manually using Osprey (Matlab based program) and calling rates used to identify spatial and temporal distribution of each call type at each location.
3. CONCLUDING REMARKS

Where possible, long term data sets and hydrophone continuity, between sites, would be ideal for clarifying distribution limits and identifying migration routes and temporal patterns. Passive acoustics monitoring is a non-intrusive and economical tool to study and monitor visually elusive vocal taxa over extended periods. Knowing the variability of a species in time and space is important for appropriate and up to date species management and conservation.

4. ACKNOWLEDGEMENTS

Data for the Perth Canyon, Bass Strait and Tasman were sourced from the Integrated Marine Observing System - an initiative of the Australian Government being conducted as part of the National Collaborative Research Infrastructure Strategy and the Super Science Initiative.

REFERENCES


Fig.1: Hydrophone deployment positions off the southeast Indian Ocean and southwest Pacific Ocean.


Abstract: Passive acoustic monitoring (PAM) is widely in use to study marine mammals in their underwater habitats. Since marine mammals can be found in all ocean basins, their habitats cover diverse underwater environments. Properties of the ocean environment, such as sound speed profile, bathymetry, and sediment properties can be markedly different between these diverse habitats, leading to differences in how a marine mammal’s vocalization is altered by propagation effects. This distortion of vocalizations may impact the accuracy of PAM systems. Thus, to develop a PAM system capable of operating in numerous environments one must understand how propagation effects impact these systems.

Previous effort has shown that a prototype aural classifier developed at Defence R&D Canada could successfully discriminate several cetacean species’ vocalizations in a relatively limited data set. The aural classifier was found to be an effective PAM tool because it employs perceptual signal features, which model features used by the human auditory system. The current work used the OASES (Ocean Acoustics and Seismic Exploration Synthesis) pulse propagation model to examine the robustness of the classifier under various environmental conditions. Preliminary results from transmitting cetacean vocalizations over several ranges in a simulated underwater environment are discussed. The modelled environment used to obtain these results was based on environmental data collected during propagation trials. Aural classification accuracy was compared for signals propagated over different ranges and provided a preliminary measure for the robustness of the perceptual features to propagation effects.

Keywords: Aural classifier, marine mammals, passive acoustic monitoring, propagation
1. INTRODUCTION

Properties of the ocean environment – such as sound speed profile (SSP), bathymetry, sediment properties, and ambient noise profiles – can be markedly different between regions where passive acoustic monitoring (PAM) is used to observe cetaceans. This leads to differences in sound propagation characteristics and results in distortion of received cetacean vocalizations [1]. The accuracy of PAM systems may be affected if this is not accounted for; however, little research has yet been directed towards this problem. Helble et al. [2] demonstrate the significant impact of the ocean environment on PAM. By using a parabolic equation propagation model to simulate calls originating from sources at various locations with respect to a fixed receiver, they were able to show that the probability of detecting a humpback whale call is environment-dependent and can be markedly different between monitoring locations. Based on these results, it may be concluded that a thorough understanding of how the environment impacts the signal features used for detection and/or classification is required to develop an automatic recognition system capable of operating effectively under numerous environmental conditions.

Previous results show that a prototype computer-based aural classifier developed at Defence Research and Development Canada (DRDC) [3] can be used to successfully discriminate vocalizations from several cetacean species [4]. The success of the aural classifier is due to the perceptual signal features it employs. These are different than features obtained using conventional signal processing techniques because they take into account how a listener perceives sound [3, 4].

The current work aims to determine the robustness of the aural classifier under various environmental conditions through a combination of empirical measurements and simulated results. This paper focuses on results obtained by using the Ocean Acoustics and Seismic Exploration Synthesis (OASES) propagation model [5] to simulate synthetic bowhead and humpback vocalizations propagated over ranges of 0 to 20 km. The properties used to characterize the modelled environment were based on environmental data collected during a propagation experiment that took place in the Gulf of Mexico during the spring of 2013.

2. DATASET

Recordings of example bowhead and humpback vocalizations were obtained from the MobySound website [6]; however, these calls were subject to unknown propagation effects when they were recorded. Therefore, synthetic signals were developed to provide known starting signals with no propagation effects applied prior to the experiments. The synthetic signals were based on an example set of high signal-to-noise ratio recordings of bowhead song endnotes and humpback song units. To generate the synthetic signals, the noise was reduced in each example call using wavelet analysis. Then the mean signal and empirical orthogonal functions (EOFs) were calculated from the wavelet-transformed vocalizations for each species [7]. To generate a single synthetic signal, random weights were applied to the EOFs that contained 95% of the variance, the weighted EOFs were added to the mean signal, and finally the inverse wavelet transform was performed. Applying different randomized weights to the EOFs for each synthetic signal generated a set of 155 signals per species. The resulting collection of synthetic signals had similar mean and variance for the perceptual features that were considered most important for
Fig. 1: Example spectrograms of (a) bowhead, (b) humpback, (c) synthetic bowhead, and (d) synthetic humpback vocalizations.

discriminating between the example bowhead and humpback vocalizations. Fig. 1 shows example spectrograms of the real and synthetic bowhead and humpback vocalizations.

3. METHODS

A two-day sea trial was conducted in the Gulf of Mexico, approximately 74 km south of Panama City, FL, from 30 April to 1 May 2013. A pictorial summary of the experimental setup is given in Fig. 2 (a). Two moorings were deployed, each with two hydrophones at different depths within the water column. Then both real and synthetic vocalizations were transmitted to the moored receivers from a source deployed from CFAV QUEST while the ship drifted. After the transmissions were completed (approximately one hour), the range from the moored receivers was increased and the transmissions were repeated. Environmental properties were measured throughout the experiments in order to understand the propagation conditions, and to provide realistic parameters for modelling. CTD casts were performed at each location the signals were transmitted to characterize water column properties. Information on the sediment characteristics was obtained from several Free Fall Cone Penetrometer (FFCPt) casts along the ship’s track [8]. Data from the FFCPt provided information on the sediment type (e.g., silty-clay, sand), from which the relevant geo-acoustic parameters were estimated.

To simulate this experimental procedure, the pulse propagation module of OASES [5], referred to as OASES-OASP, was used to propagate signals through a range-independent environment. OASES is a general-purpose computer code that uses the wavenumber integration method for modelling seismoacoustic propagation in horizontally stratified waveguides. OASES-OASP calculates the depth-dependent Green’s function and determines the acoustic transfer function at each receiver by evaluating the wavenumber integral [5]. The model outputs the received pressure time series for each source signal.

The signals propagated with OASES-OASP were input to the aural classifier algorithm (note that the signals recorded during the experiments are not considered in this paper). The aural classification process is broken into three phases: First, a simple auditory model is applied. Second, the perceptual signal features are calculated for each signal. Finally, a Bayesian classifier is applied [3, 4]. The classifier was trained with signals propagated
Fig. 2: (a) Representation of the experimental setup. The ship first deployed two hydrophone moorings, moved to the first location and transmitted the set of signals, then moved further away from the recorders and retransmitted the signals. R₁ and R₂ represent the horizontal range the signals propagated from source to the midpoint between the moorings. (b) Sound speed profile measured during the morning of 30 April 2013. This was used to characterize the water column for modeling with OASES-OASP.

This classifier was then tested with signals propagated in the model over 5, 10 and 20 km ranges, for which the classifier had no direct knowledge of the associated class labels. Classification accuracy and area under the ROC curve, AUC, were used to evaluate classifier performance. The AUC can vary between a value of 1.00, indicative of an ideal classifier, and 0.50, equivalent to randomly assigning a classification decision. In this way, OASES-OASP was used in conjunction with the aural classifier to study the effects of propagation on the perceptual features as a function of range between source and receiver.

4. RESULTS AND DISCUSSION

The model was run for geometries and environmental parameters consistent with the Gulf of Mexico sea trial. The environment was modeled as a water column overlaying a sediment half-space. The geo-acoustic parameters of the sediment half-space (density = 1.77 g/cm³, compressional sound speed = 1646 m/s, shear sound speed = 400 m/s, compressional attenuation = 0.8 dB/λ, and shear attenuation = 2.5 dB/λ) were consistent with the silty-clay sediment type observed at the experimental site. The sound speed profile shown in Fig. 2 (b) was measured the morning of 30 April 2013 and was used to model the water column characteristics. The following results considered only propagation of the synthetic signals (real vocalizations will be analyzed in future work) with the source and receiver located at depths of 30 m and 36 m, respectively.

Fig. 3(a) shows results obtained by training the classifier on synthetic signals propagated with OASES-OASP over a range of 0 km. The remaining panels in Fig. 3 depict the results of testing the classifier on signals propagated with the model over ranges of 5 to 20 km. These plots show histograms of projected feature values that were obtained from a linear combination of the five features (duration, global mean subband decay time,
Fig. 3: Classification results for synthetic bowhead (blue) and humpback (red) signals. (a) The classifier was trained using signals that were propagated using OASES-OASP over 0 km range. The dashed lines in panel (a) represent the theoretical Gaussian PDFs used to determine the decision regions. The classifier was tested on signals that were propagated over ranges of (b) 5 km, (c) 10 km, and (d) 20 km.

Table 1: Accuracy, AUC, and class means and variances for the synthetic vocalizations.

<table>
<thead>
<tr>
<th>Transmission Range [km]</th>
<th>Accuracy</th>
<th>$AUC$ (Bowhead)</th>
<th>Variance (Bowhead)</th>
<th>Mean (Humpback)</th>
<th>Variance (Humpback)</th>
</tr>
</thead>
<tbody>
<tr>
<td>0</td>
<td>100%</td>
<td>1.00</td>
<td>1.70</td>
<td>-1.70</td>
<td>0.09</td>
</tr>
<tr>
<td>5</td>
<td>91%</td>
<td>-0.13</td>
<td>1.98</td>
<td>-3.01</td>
<td>0.33</td>
</tr>
<tr>
<td>10</td>
<td>92%</td>
<td>-0.01</td>
<td>1.28</td>
<td>-3.22</td>
<td>0.17</td>
</tr>
<tr>
<td>20</td>
<td>88%</td>
<td>0.97</td>
<td>-0.03</td>
<td>1.01</td>
<td>-3.04</td>
</tr>
</tbody>
</table>

local maximum subband decay time, frequency of global maximum subband attack time, and peak loudness value [3, 4]) that best discriminated between the real vocalizations from MobySound. Correct classification is represented by a coloured histogram bin plotted on the background of the corresponding colour (e.g., dark blue on light blue). Quantitative results are listed in Table 1.

The classifier was able to discriminate between the signals in the training set with 100% accuracy and $AUC$ of 1.00. There was a trend for accuracy to decrease as the propagation range increased, although the $AUC$ values showed little change. The $AUC$ values were similar because the separation between the bowhead and humpback classes was maintained. These two trends are reflected in the decision regions in panels (b) through (e) of Fig. 3; in each there was little overlap between class distributions, though the distributions shifted with respect to the decision threshold (i.e., the boundary separating the red and blue backgrounds). The variance of both the bowhead and humpback classes changed with propagation range. These preliminary results suggested
that propagation conditions affect at least one of the five perceptual signal features used for classification.

5. CONCLUSIONS

The OASES-OASP propagation model was used to propagate synthetic signals through a modelled environment based on environmental properties measured during a sea trial performed in the Gulf of Mexico. Preliminary classification results indicate that the environment may impact some of the perceptual signal features. Further work is required to quantify how robust each of the perceptual features is to propagation effects and which environmental properties most affect the features. Future work also includes comparing results from propagation models to results obtained from the sea trials.

6. ACKNOWLEDGEMENTS

The authors would like to thank Dr. Sean Pecknold and Mr. Stefan Murphy of DRDC, and Dr. Tetjana Ross of Dalhousie University for advice and insights. The authors wish to acknowledge MobySound for the availability of marine mammal vocalization data. This work was supported in part by the U.S. Office of Naval Research, Code 32, Marine Mammals & Biological Oceanography Program under Grant N00014-14-1-0237.

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CLASSIFICATION OF BEAKED-WHALE SIGNALS RECORDED IN ATLANTIC

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Abstract: Beaked whales are a group of more than twenty genetically confirmed species; they are very elusive and were among the least known species until a few years ago. Because of their sensitivity to sonar, an increased research effort dedicated to these species started ten years ago. Fourteen different types of beaked whale echolocation clicks are known, nine of which attributed to a species and the other five not yet assigned. All of the signals are up-sweep frequency modulated and seem to be species specific. In 2010, NATO Undersea Research Centre (NURC) conducted a sea trial in Eastern Atlantic Ocean, Southwest of Portugal. Three different types of beaked whale signals were recorded. Two of them have been assigned to a species. The third type has similarities with Cuvier’s beaked whale signals but also exhibits some discrepancies with what is typically found in the literature. The main signal characteristics (mean spectrum, Inter-Click Interval histogram,...) are compared with those coming from Cuvier’s beaked signals recorded with the same acoustic device in the Mediterranean Sea. The results of the comparison are presented.

Keywords: Beaked Whale, Classification
1. INTRODUCTION

Description of acoustic signals have been made for Arnoux’s (Berardius arnuxii) [1], Cuvier’s (Ziphius cavirostris) [2,3], Blainville’s (Mesoplodon densirostris) [2,4], Gervais (Mesoplodon europaeus) [5], Deraniyagala’s (Mesoplodon hotaula) [6], Longman’s (Indopacetus pacificus) [7], Northern bottlenose Whale (Hyperoodon ampullatus) [8], likewise Stejneger’s (Mesoplodon stejnegeri) [9], Baird’s (Berardius bairdii) [10], and Sowerby’s (Mesoplodon bidens) [11]. In addition, five distinct pulse types from unknown species have been described [12].

Beaked whale echolocation clicks are mostly frequency modulated upsweep and appear to be species specific [12]. In [12], the authors compare the signals of twelve species, seven of which are identified and five come from unknown species. As all toothed whale clicks, beaked whale clicks are highly directional [3, 13]. Most of the papers only described on-axis clicks; the particularity of [12] is that all clicks (regardless of distance and orientation of the animals to the recorder) are used for the analysis. This paper compares all known beaked whale echolocation clicks, except Sowerby’s beaked whale, Northern Bottlenose whale and Arnoux’s beaked whale.

In 2010 NATO Undersea Research Centre (NURC, now CMRE Centre for Maritime Research and Experimentation) conducted a sea trial in Atlantic Ocean, called Sirena10. During this cruise, three different types of beaked whale signals were recorded. The recorded signals were compared to all known beaked whale signals [8, 11-12] except Arnoux’s beaked whale signals because they don’t live in this area. Two of them have been assigned to a species. The third type clicks, called Type3 in this document, have similarities with Cuvier’s beaked whale signals analyzed in [12], hereafter called CUV1, but also exhibit some discrepancies. Some observed differences may be due to differences in recording system, e.g. intrinsic characteristics and deployment depth. It has been noted in [6, 10] that the same species of beaked whale signals look different when recorded from a towed array or from a bottom-mounted device. The signals were recorded by the former in Sirena 10 whereas they were recorded by the latter in [12]. Cuvier’s beaked whale signals have been recorded in the Mediterranean Sea during the 2011 NURC sea trial, called Sirena 11, by the same acoustic device as in Sirena 2010. These Sirena 11 clicks will now be called CUV2. The main signal characteristics are compared. The results of the comparison are presented in this paper.

2. METHODS

2.1. Data collection

The SIRENA 10 [14] and SIRENA 11 sea trials were performed on the NATO Research Vessel “Alliance”. The first one took place in Eastern Atlantic Ocean, off the Southwest coast of Portugal, Spain and Northwest of Morocco (Fig. 1 A), the second one in the Gulf of Genoa in the Mediterranean Sea (Fig. 1B). During both trials acoustic data were collected with the CPAM (Compact Passive Acoustic Monitoring), designed by NURC [15]. The total usable bandwidth is up to 80 kHz.

The CPAM was deployed at a depth between 100 and 200 m for about 20 hours a day. There was always one operator monitoring the real-time spectrogram and taking notes on
acoustic marine mammal encounters. Three different classes were offered to the operator: sperm whale, beaked whale or dolphins-like signals.

![A Sirena 10](image1)

![B: Sirena 11](image2)

**Fig. 1: A: SIRENA 10 survey area B: SIRENA 11 survey area**

### 2.1.1. Signal Processing

Individual clicks were automatically detected using the transient detector described in [16]. This detector uses a Page test [17]. The detector provides click start and stop times. Each beaked whale click was then manually validated by a trained operator. Recorded sequences comprising a mix of beaked whale clicks and numerous dolphin clicks were discarded for this study to prevent false classifications.

In order to compare the recorded clicks to the other known beaked whale ones, the data were processed almost the same way as in [12].

A 10-pole Butterworth band-pass filter between 5 and 95 kHz was then applied. Filtering was done on 800 samples centered on the echolocation signal. The spectrum of each detected click was calculated on aforementioned duration of the click with a Hanning window. The peak and center frequencies were processed using methods from [13]. Mean spectrum of all clicks was plotted against the mean noise preceding each click (same duration than the click with a 1.3 ms gap before the signal, bandpass filtered like the clicks).

Individual time series with normalized amplitude and spectrograms (2ms data, 60-point DFT, 98% overlap) of one of the highest amplitude click (in order to be as close as possible of an on-axis click) are plotted (Fig.2). These plots will be checked against similar plots presented in [12].

In [12] each click spectrum was calculated for a fixed duration of 2.56 ms of data centered on the click. In this work the DFT was only performed on the click itself because the signal recorded from a towed body is much noisier than data recorded in [12] from a fixed bottom-mounted sensor.

### 3. RESULTS

Fig. 2 A and Fig. 3A give Type3 characteristics. Fig. 2B and Fig. 3B give those of CUV2. Fig. 4 gives the mean spectrum of Type3 CUV1 and CUV2. Table 1 gives the characteristics of Type3, CUV1 and CUV2.
Fig. 2: Example of Time series (top) and spectrogram (bottom) of an on-axis click

Fig. 3: (Top) Mean spectrum (solid line) and mean noise preceding each signal (dashed line), (bottom) histograms of inter-click interval

Fig 4: Mean spectrum of Type3, CUV2 and CUV1 (courtesy of S. Baumann Pickering)

<table>
<thead>
<tr>
<th>Signal</th>
<th>Peak freq. (kHz)</th>
<th>Centre freq. (kHz)</th>
<th>ICI (ms)</th>
<th>Nb of clicks</th>
</tr>
</thead>
<tbody>
<tr>
<td>Type3</td>
<td>30.0 (17, 39)</td>
<td>31.3 (25, 37)</td>
<td>598 (474, 780)</td>
<td>567</td>
</tr>
<tr>
<td>CUV1</td>
<td>40.2 (20, 49)</td>
<td>35.9 (29, 42)</td>
<td>337 (94, 491)</td>
<td>46629</td>
</tr>
<tr>
<td>CUV2</td>
<td>24.4 (17, 36)</td>
<td>28.0 (22, 34)</td>
<td>471 (375, 536)</td>
<td>466</td>
</tr>
</tbody>
</table>

Table 1: Median values of Peak and center frequency, ICI, with 10th and 90th percentile in parentheses. The last column gives the number of clicks used for the analysis.
4. DISCUSSION

The on-axis click spectrogram of Type3 (Fig. 2A) seems fuzzier than the example of CUV1 given in [12], but it looks pretty similar to a CUV2 example (Fig. 2B).

Type3 mean spectrum is very similar to those of CUV1 and CUV2, with same secondary peak frequencies and notches (Fig.4). The characteristics given in Table I indicate that the Type3’ peak and center frequency are slightly different from those of the two other cases. This is likely due to the differences in depth recorder as already noted in [6, 10] and the fact that the differences of levels between the various peak frequencies are less than 4 dB. This analysis tends to make us believe that Type3 comes from Cuvier’s beaked whale.

A doubt persists because Type3’s ICI is significantly different from CUV1’s and CUV2’s. The difference between CUV1 and CUV2 is mainly due to the fact that more than one animal can be detected at the same time for the former, which causes a decrease in the ICI, whereas only one animal is detected most of the time for the latter. In [3] the ICI has been calculated for two Cuvier’s beaked whale recorded in the Mediterranean sea (on more than 10 000 clicks for each whale). The median values of ICI for these two whales were 390 ms and 410 ms which is not far from what has been calculated for CUV2 (471 ms). The Atlantic recordings have a median ICI very close to 600 ms, this seems high for a Cuvier’s beaked whale compared to what can be found in the literature.

5. ACKNOWLEDGEMENTS

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REFERENCES


Session 4

Acoustic Tomography in Shallow Seas, Benthic and Terrestrial Waters

Organizers: Jean-Pierre Hermand, Arata Kaneko and Hiroyuki Hachiya
A COASTAL ACOUSTIC TOMOGRAPHY EXPERIMENT FOR TIDAL CURRENT MEASUREMENT IN THE MOUTH OF JIAOZHOU BAY

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Abstract: A coastal acoustic tomography experiment for measuring the tidal current in the mouth of Jiaozhou Bay on the western coast of the Yellow Sea was carried out with three acoustic stations in September, 2010. The carrier of frequency 5 kHz, modulated with the 10\textsuperscript{th} order M-sequence was transmitted every 3 minutes, and the reciprocal sound transmission was successfully among the three stations. During the sound transmission experiments, twenty-eight repeat shipboard Acoustic Doppler Current Profiler (ADCP) surveys were also performed along the sound transmission line to get a comparison with the reciprocal sound transmission data. Based on the result of ray simulation, the received signals are divided into two groups which travel through the top 15 m layer and full depth water layer, respectively. An inversion with regularization is applied to estimate current velocity in the upper and lower layer using the differential travel times from the two ray groups. The average current velocities along the vertical section in the upper and lower layer, determined by the inverse analysis using the travel time differences, were in good agreement with the ADCP results. The root-mean-square differences between two measurements in the upper and lower layer are 4.6 cm s\textsuperscript{-1} and 5.8 cm s\textsuperscript{-1}, respectively.

Keywords: Coastal acoustic tomography, tidal current, inversion, Jiaozhou Bay.
1. INTRODUCTION

The continuous monitoring of tidal currents in bays, inland seas and straits is a difficult, especially in the coastal seas of China, where fishing and shipping pressures are high. To obtain observations of rapidly varying tidal current structures, bottom-mounted or subsurface moored current meter arrays are needed as conventional tools. However, heavy shipping traffic and fishery activities make it difficult to deploy such mooring arrays in almost all coastal regions in China.

Ocean acoustic tomography (OAT) was proposed by Munk and Wunsch in 1979 as a powerful oceanographic technique for mapping mesoscale oceanic phenomena [1]. Coastal acoustic tomography (CAT) was proposed as an application of OAT to the coastal sea, aiming at the continuous monitoring of tidal currents in ports, bays, straits and inland seas without disturbing shipping traffic, fisheries and marine aquaculture activities [2].

The CAT system, developed by Hiroshima University, has been successfully applied to current structural measurement in coastal seas around Japan since 1995 [3]-[5], and applied recently in China to invert the horizontal distribution of tidal current [6] and river discharge with tidal bores [7]. This paper reports a CAT experiment that was executed in the mouth of Jiaozhou Bay on the western coast of the Yellow Sea, China.

2. EXPERIMENT SITE AND DATA COLLECTION

Jiaozhou bay (JZB), located on the western coast of the Yellow Sea, is a semi-closed shallow bay with an area of about 380 km², an average water depth of 7 m and a maximum depth of about 60 m at the bay mouth (Fig.1). JZB is connected to the Yellow Sea by a narrow channel. The width and mean water depth of this channel are approximately 3 km and 25 m, respectively.

A coastal acoustic tomography experiment with three CAT systems was performed during September 19-21, 26 and 29-30, 2010, in the mouth of JZB (Fig.2). The three CAT systems were set up at three stations numbered A, B and C using fishing boats anchored on the two sides of the channel. The transducer was suspended down to 5 m depths from the fishing boat by a rope, while the major components of the system, such as electronic housing, batteries and GPS antenna were put onboard the boat.

A 5 kHz sound with a bandwidth of 5 kHz/3=1.7 kHz, modulated by one period (1024 digits and 0.64 s) of the 10th order M sequence, was transmitted every 3 minutes simultaneously from each broadband transducer (Neptune T170) to increase the signal-to-noise ratio (SNR) of the received signals by $20 \log \sqrt{2^{10} - 1} = 30.1$ dB. M sequences with different codes were assigned for each of the acoustic stations to identify arrival signals from the multiple stations. A one-period M sequence (0.64 s) was transmitted from the three stations with good timing accuracy synchronized by GPS. By using the different M sequence codes, acoustic signals traveling to one station from the other stations are resolved with the time resolution for multi-path arrival (0.6 ms), defined as one digit of the M sequence.

The shipboard acoustic Doppler current profiler (ADCP, RDI 300-kHz) observations were performed in the daytime along the three transmission lines, at a schedule synchronized to the CAT experiment. Total of 28 sections (21 for section AB, 2 for section BC and 5 for section AC) shipboard ADCP data were collected for comparison with the CAT data. The ADCP was set up at the side of a wooden fishing ship using a
stainless steel frame and its transducers were maintained at 1 m depth. The ship speed during the survey was kept at 3-5 m s\(^{-1}\). Three CTD casts were executed during the shipboard ADCP surveys to determine the sound speed profile.

Fig.1: Location maps of the experimental site. The positions of the CAT stations (A, B and C) are shown with circles. The black lines connecting the CAT stations indicate the sound transmission lines. The interval of bathymetric contours is 10 m.

Fig.2: Tidal level anomaly (grey lines) obtained inside of Jiaozhou Bay near the Qingdao city. The time schedules of the CAT, shipboard ADCP and CTD surveys are indicated by the dots, horizontal bars and vertical lines, respectively. The cruise number of shipboard ADCP is shown above or under the bars.
3. RESULTS

3.1 Ray simulation

Fig. 3 shows the ray pattern obtained between all the three station pairs by the ray-tracing method using the range-averaged sound-speed profiles determined from the CTD profile data and the sea bottom topography obtained from shipboard ADCP measurements using the bottom track mode. The sound speed increased from surface to bottom (Fig. 3 (a1, b1 and c1)), which caused the ray path undergoing 3-4 bounces at surface (Fig. 3 (a2, b2 and c2)). The ray lines crossed almost all the depths of the three sections except the deepest trough in section AC (Fig. 3 (c2)).

To examine the relationship between travel time and ray pattern, the travel time is plotted against depth (right panels of Fig. 3). The ray lines between Section BC can be derived into two groups. Group I: the ray lines distribute only in the top 15 m layer (the red lines in Fig. 3 (a2)), which travel time is shorter than 1.867 second (the red dots in Fig. 3 (a3)). Group II: the ray lines cover the all section with several bounces between surface and bottom (the blue lines in Fig. 3 (a2)), which travel time is between 1.868 and 1.869 second (the blue dots in Fig. 3 (a3)). The travel times of ray line for Section AC and Section BC have a very short range, and those ray paths could not be grouped.

![Fig. 3 Ray patterns obtained between the three stations by the ray-tracing method using the range-averaged sound speed profiles. The plots of mean travel time between the station pairs against depths are shown in the right panels. The upper, middle and bottom panels are for the Section AB, AC and BC, respectively.](image)

3.2 CAT Data

Reciprocal sound transmissions were measured along the 3 transmission lines. The correlation waveforms of the received signals between A–B are shown as an example in units of signal-to-noise ratio (SNR) with the stack diagrams in Fig. 4. The correlation
waveform formed a steep arrival peak with large SNRs for the first arrival peak (black dots), while the second arrival peak (red dots) is much lower than the first one. The travel time is determined at the peak point corresponding to the maximum SNR both for the first and second arrival. Then the differential travel times for the ray Group I and Group II are obtained by subtracting the travel times of first and second arrival, respectively.

![Image](image.png)

Fig.4: Arrival patterns stacked upward with time. The correlation waveforms received at A from B are shown in right panel, while that received at B from A are shown in left panel. The black and red dots are put on the peaks that belong to ray Groups I and Groups II, respectively. Note that the first arrival peak is aligned with zero travel time.

3.3 Shipboard ADCP Data

The vector plots of depth-averaged current velocity along the sound transmission lines obtained by shipboard ADCP from 28 cruises are shown in Fig.5, with the time corresponding to $M_2$ phase. During the beginning of flood, the current is eastward, flowed into JZB in the northern of the channel, while the current is westward in the southern of the channel (AB1-AB2 in Fig.5). Then the current gradually turns to westward at the southern side of the channel (AB3-AB5 in Fig.5), and the westward current along the Section AB reaches its maximum (AB6-AB8 in Fig.5). During the transition phase from flood to ebb, the current is weaker at southern than that at northern of the channel (AB9 in Fig.5). During the ebb tidal period, the eastward current is stronger at southern than that at northern of the channel (AB12-AB21 in Fig.5).

3.4 Inverse Results

Following the ray grouping method proposed by Taniguchi et al. [8], we obtained two groups of the travel time difference data, which correspond to the first and second arrival. The inversion with regularization method [9] is used to estimate the range-averaged current velocity ($V_{CAT}$) at surface layer (0-15 m) and bottom layer (15 m–) for Section AB estimated by the inverse method (Fig.6(a)) and for Section BC and Section AC for the full depth (Fig.6(b)-(c)). $V_{CAT}$ is also compared with the shipboard ADCP velocities ($V_{ADCP}$).
averaged for the whole transect. $V_{CAT}$ was in good agreement with $V_{ADCP}$, producing a root-mean-square difference (RMSD) of about 0.02-0.05 m s$^{-1}$.

Using the $V_{CAT}$ along the three CAT transect lines, the current vectors at four sub-triangles are inverted following a method proposed by Huang et al. [10]. The major axis of the tidal ellipses for M2 tidal current was directed east-west near Station B, and turned to northward near Section AC (Fig. 7). The maximum velocities of the M2 tidal current at the four sub-triangles reached 0.90 m s$^{-1}$ and 0.64 m s$^{-1}$ for upper and lower layer, respectively. The residual currents are small with the velocities about 0.03-0.05 m s$^{-1}$ toward the south.

![Vector plots of depth-averaged current velocity along the sound transmission line obtained by shipboard ADCP. The tidal phases are indicated by a dot on the sea level anomaly plot at the upper of each panel.](image)

4. SUMMARY

The coastal acoustic tomography experiment for measuring the tidal current in the mouth of JZB on the western coast of the Yellow Sea was successfully carried out with three acoustic stations during September 19-21, 26 and 29-30, 2010. The 10th order M-sequence was transmitted every 3 minutes, and the reciprocal sound transmission was successfully among the three stations. During the sound transmission experiments, repeat ADCP surveys were also performed along the sound transmission line.

Based on the ray simulation result, the received signals are divided into two groups which travel through the top 15 m layer and full depth water layer. An inversion is adopted to estimate current velocity in the upper and lower layer using the differential
travel times from the two ray groups. The average current velocities along the vertical section in the upper and lower layer, determined by the inverse analysis using the travel time differences, were in good agreement with the ADCP results. The root-mean-square differences between two measurements in the upper and lower layer are 4.6 cm s\(^{-1}\) and 5.8 cm s\(^{-1}\), respectively. The major axis of the tidal ellipses for \(M_2\) tidal current was east-westward near Station B, and turned to northward near Section AC, with a maximum velocity of 0.90 m s\(^{-1}\) and 0.64 m s\(^{-1}\) at upper and lower layer, respectively. The residual currents are small with the velocities of about 0.03-0.05 m s\(^{-1}\) toward the south.

5. ACKNOWLEDGEMENTS

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Fig. 6: (a) Time series of range-averaged current velocity estimated by the inverse method for Section AB. The blue and red lines are for the depth-averaged in the top 15 m and below 15 m, respectively. The blue and red horizontal bars indicate the average ADCP velocities in the top 15 m and below 15 m, respectively. Time series of range-averaged current velocity estimated directly using travel time difference data for (b) Section BC and (c) Section AC. The black horizontal bars indicate the depth-averaged ADCP velocity from each shipboard ADCP survey.
Fig. 7: $M_2$ tidal current ellipses and residual current (vectors) estimated by a tidal harmonic analysis method using the CAT velocity data averaged in top 15 m layer (blue) and below 15 m layer (red).

REFERENCES

ACOUSTIC INVESTIGATIONS OF UNSTEADY SALINITY INTRUSION IN A DIVERSION CHANNEL

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Abstract: Long-term continuous measurements of tidal current and sound speed/salinity have been conducted in a tidal diversion channel using Fluvial Acoustic Tomography System (FATS) with a couple of 30 kHz broad-band transducers. The FATS was located around 8.7 km upstream far from the mouth. The reciprocal sound transmission that was performed between the two acoustic stations, located on both sides of the channel, enabled us to measure range-averaged sound speed and water velocity along a ray path. The channel is a shallow tidal-forced river with a maximum tidal range of 4 m at the mouth. The tides are primarily semidiurnal, but mixed with a diurnal component. The freshwater runoff into the diversion channel was regulated by the array of sluice gates, located 270 m upstream of the observation site. Although only one sluice gate was usually opened slightly, all sluice gates were completely opened during flood events. The saline water was flushed out by the gate operation for flood events. Thus, the salinity intrusion in the channel presented significant unsteady nature. Since the salinity varied in the span of 0 to 25 owing to the tides, the sound speed was significantly influenced by the salinity. The tidal velocity amplitude (\(U_T\)) and the outflow velocity (\(U_R\)) associated the river discharge controlled the salinity intrusion. The recovery of salinity intrusion after the gates were set at the normal condition (slightly opened) was hardly found when \((U_T U_R)^{-1}\) was smaller than around 60 m\(^{-2}\)s\(^2\). The recovering time of the salinity ranged from 9.5 days to 27 days.

Keywords: Saline intrusion, tidal current amplitude, river flow
1. INTRODUCTION

Adjustments to salinity intrusion may have important implications for factors such as water use/quality, sediment transport/sedimentation and dispersion of pollutants. The nature of salinity intrusion in an estuary is governed by tidal current amplitude, river flow, bathymetry, wind, and so on. In addition, the pattern of intrusion may be altered by changes in the mean sea level or river flows linked to Global Climate Change.

Although modelling for flows and transport phenomena is relatively cheap and continues to advance rapidly, even sophisticated 3D numerical models may encounter difficulties in reproducing the complexity and diversity of mixing and sedimentary processes. On the other, long-term measurements of velocity and salinity by the conventional methods are expensive and laborious.

In the present study, long-term measurements of velocity/discharge and sound speed were carried out using the fluvial acoustic tomography system (FATS) in the Ota River diversion channel with large changes of water depth and salinity. Since FATS can automatically collect the data of sound arrival time at an unmanned site, FATS saves us labor for the long-term monitoring of saline intrusion.

2. OBSERVATION SITE AND METHOD

The observations were carried out during June to September 2013 at the Ota diversion channel (Fig. 1). The Ota River bifurcates into two main branches at about 9 km upstream from the river mouth. The tidal limit in Ota River estuary is about 13 km upstream far from the mouth. River flow in this tidal compartment is characterized by the periodic intrusion of salt wedges. The tides are primarily semidiurnal, but mixed with a diurnal component. The tidal range at a spring tide can be as large as 4 m at the mouth. The discharge of the Ota River is monitored at the Yaguchi gauging station that is located at 14.2 km upstream from the mouth.

Freshwater runoff is usually limited by the Gion sluice gates that are located at the bifurcation place: 270 m upstream of the observation site. Only one sluice gate is opened slightly in order to make a cross-sectional area of stream is $32 \text{ m} \times 0.3 \text{ m}$ to spill the flow.

The observation site was located at 270 m downstream from the Gion sluice gates as shown in Fig. 1. The Ota diversion channel at the site is 120 m wide and the water depth ranges from 0.3 m to 3 m by tide. The salt water in the Ota River can intrude to about 11 km upstream from the mouth.

A couple of broadband transducers were installed diagonally across the channel as shown in Fig. 1. The central frequency and bandwidth of transducers were 30 kHz and 18 kHz, respectively, angle between sound pass and stream direction $\theta$ was 30 degrees, and transducers were mounted at the height of 0.2 m above the bottom. The altitudes of left and right transducers were $-0.46$ m above mean sea level (masl) and $-0.7$ masl, respectively. The sound pulses of the FAT system were simultaneously transmitted from the omni-directional transducers triggered every minute by a GPS clock.

Water level was measured every hour near both transducers. Vertical distribution of water temperature and salinity were measured every 10 minutes by conductivity-temperature (C-T) sensors attached to the pier of the Gion Bridge (Fig. 2) 40 m far from the left bank. The heights of respective C-T sensors were $-1$, 0, 1 masl.
3. RESULTS

3.1. Time series of river discharge, water level and flow rate at the site

Figure 2 shows time series of the river discharge at the Yaguchi gauging station $Q_Y$, the water level $H$ and the flow rate at the observation site $Q_G$. The flow rate $Q_G$ is calculated by the following formula [1]:

$$Q_G = A(H)u_m \tan \theta$$

where $A(H)$ is the cross-sectional area in which sound makes ray paths, $H$ is the water level, $u_m$ is the cross-sectional average velocity along the transmission line and $\theta$ is the angle between the ray path projected to the horizontal plane and stream axis.

The each record (June, August, and September) includes one flood event. The Gion gates are completely opened during flood events: $Q_Y \geq 400 \text{ m}^3\text{s}^{-1}$. Unfortunately, the sound propagation was interrupted by the suspended sediment or sedimentation on the transducers in the larger flood events. Since only one sluice gate of the Gion is opened slightly for the usual period, the flow rate at the observation site is small.

The tidal range at the site ranges from 0.5 m to 3 m. For the flood event on 4 September, the water level exceeds the flood plain height of 3 m.

The ratio of $Q_G$ to the river discharge $Q_Y$ was approximately 0.7 during the flood events and the maximum velocity in the main channel was around 1.5 ms$^{-1}$. The tidal current amplitude ranged from 0.06 to 0.14 ms$^{-1}$ under the normal gates condition.

3.2. Temporal variations in sound speed and salinity
The reciprocal sound transmission that was performed between the two acoustic stations, located on both sides of the channel, enable us to measure range-averaged sound speed along the ray path. Figure 3 demonstrates temporal variations in the sound speed estimated from the reciprocal sound transmissions of FATS; the effect of water velocity on sound speed is eliminated in the results shown in Fig. 3.

The sound speed is more sensitive to water temperature $T$ rather than salinity $S$. The sensitivity of sound speed to $T$ is approximately 3 times greater than that to $S$. However, the tidal salinity amplitude was around 10 times greater than the temperature amplitude at
Fig. 3: Temporal variations in the sound speed measured by FATS; blue and black arrows denote complete opening and getting back of gates; red arrows indicate the first sharp peaks of sound speed after the getting back.

Fig. 4: Temporal variations in salinities at the Gion bridge pier; the red line denotes salinity from CT1 at -1 m above mean sea level (masl), blue: 0 masl, black: 1 masl.
the observation site. Thus, the semidiurnal fluctuations in the sound speed are mainly caused by the periodical salinity intrusion.

As shown in Fig. 3, the semidiurnal fluctuations are diminished by the gates opening because the salt water is flushed out. The red arrows denote the re-intrusions of salinity suggested by the sound speed fluctuations. It seems that the recovering time of the salinity ranges from 9.5 days to 27 days. Moreover, the amplitude of the semidiurnal fluctuations changes over the cycle of spring–neap tides: the amplitude during the neap tides is greater than that during the spring tides.

Figure 4 presents salinity data measured every 10 minutes by the conductivity-temperature (C-T) sensors, attached to the pier of the Gion Bridge at 40 m from the left bank (Fig. 1). The red arrows denote the re-intrusions of salinity suggested by the sound speed fluctuations (Fig. 3). The salinity behaviour found from C-T sensors differs slightly from that from the sound speed. Namely, the salinity peaks in the C-Ts time series for (24 June–3 July; 1–3 July; 28–30 August; 10–14 September) are not found from the sound speed records. It seems that the salinity effect does not emerge in the sound speed measured by FATS when the salinity is limited in the lower layer.

The differences are probably caused by the difference of sampling points between the FATS and C-T sensors. FATS estimates link with the range averaged salinity along ray paths. As discussed below, the anomaly, caused by the inability of sound paths to penetrate into the bottom layers, is a consequence of salt water intrusion to a depth below the transducer level [1, 2].

4. DISCUSSION

Typical rays and speed of sound are shown in Fig. 5 (a), (b) [2]. The typical result of ray tracing for 20:30 JST shows a pattern in which rays are entirely trapped in the upper 0.8 m layer containing fresher and colder water that leads to slower sound speed. Figure 5 (c) shows stack plots of arrival time for the signals transmitted from the right station and
Fig. 6: Temporal variations in $(U_I U_R)^{-1}$; dashed horizontal arrow denotes the periods when salinity is detected by only C-T sensors.

received at the left station. Arrival time is determined by taking the cross correlation between the M-sequence used in the transmission and the received signal. The sudden jump in the peak position of correlation might be attributed to the moment when transducers were submerged completely into the salt wedge.

In the case of Fig. 5 (a), FATS cannot detect salinity because sound paths penetrate into the salinity layers. However, the undetected period is limited to 3 hours on 10th August. Thus, the peak of sound speed is observed on 10th August as shown in Fig. 3. Therefore, the salinity may be localized during the periods when the salinity detected by only the C-Ts.

The total upstream and downstream excursion for a semi-diurnal tide is approximately 6 km in the spring tide. Therefore, the tidal advection cannot carry a fluid column to the observation point: the salt water flushed out is not able to reach the site by the tidal advection. The gates condition introduces a further ‘degree of freedom’ in determining the axial location and the length of the intrusion.

The recovering time of the salinity appears to be independent on the flood discharge as shown in Figs. 2–4. The re-intrusions of salinity in July and September are observed in the neap tides because of the weak mixing. Prandle [3] proposed the following expression for saline intrusion length $L_I$:

$$L_I = \frac{k D^2}{f U_I U_R}$$  \hspace{1cm} (2)

where $f$ is bed friction coefficient (≈ 0.0025), $U_I$ tidal current amplitude, $U_R$ river flow, $D$ mean water depth. It is widely observed that intrusion lengths in stratified conditions are significantly longer than in mixed. For a well-mixed or salt wedge type, the coefficient $k$ is 0.011 or 0.07.
Figure 7 demonstrates the temporal variations in \((U_1 U_R)^{-1}\), where \(U_R\) is the residual component of the cross-sectional average velocity measured by FATS. The saline water scarcely able to reach the observation site when \((U_1 U_R)^{-1}\) is smaller than around 60 m\(^{-2}\) s\(^2\). Since the minimum tidal current amplitude is 0.06 m s\(^{-1}\) in the neap tide, the outflow velocity \(U_R\) greater than around 0.3 m s\(^{-1}\) prevents the saline intrusion to the site over the cycles of Spring–Neap tides. The coefficient \(k \approx 0.11\) is fairly larger than the previous results [3].

5. CONCLUSIONS

The fluvial acoustic tomography system (FATS) applied to a shallow tidal estuary with periodic saltwater intrusion. The FATS composed of a couple of transducers, which are installed diagonally across the channel, was able to measure the cross-sectional average velocity and sound speed. The behaviour of unsteady saline intrusion was investigated from variations of the sound speed and the flow velocity measured by FATS.

Although the freshwater runoff was usually limited by an array of sluice gates that is located 270 m upstream of the observation site, the gates were completely opened during flood events. The semidiurnal fluctuations of salinity were diminished owing to the gates opening because the salt water was flushed out. The re-intrusions of salinity were investigated by the sound speed fluctuations. The recovering time of the salinity ranged from 9.5 days to 27 days. The recovering time was independent on the flood discharge. It was found that the saline water scarcely able to reach the observation site when \((U_1 U_R)^{-1}\) is smaller than around 60 m\(^{-2}\) s\(^2\).

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VERTICAL PROFILING OF TEMPERATURE AND VELOCITY FROM THE QUITE LIMITED DATA SET OF COASTAL ACOUSTIC TOMOGRAPHY

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Abstract: The vertical profile reconstruction of temperature and velocity is applied to two-station coastal acoustic tomography data, obtained in September 2012 in the Akinada of the Seto Inland Sea, Japan. Only two arrival peaks are identified in the correlation waveforms of received acoustic data because of the mean floor depth 35 m and the station-to-station distance 13.769 km. The ray paths corresponding to the two arrival peaks with a travel-time difference of 0.005 s are determined by the range-independent ray simulation based on the observed sound speed data. The path-averaged temperature and velocity along the two ray paths are converted into the range-averaged temperature and velocity for the five horizontal layers (0-5m, 5-10m, 10-15m, 15-20m and 20-50m) by the inversion with three-point regularization, accompanied by the Lagrange multiplier. The two-day low-pass filtered temperature reconstructed is in good agreement with the monthly observed temperature. The vertical profiles of tidally oscillatory flow are also reconstructed during half the M2 period on September 14. It is found from the power spectral analysis that the 3-day oscillation of velocity occurs even in the homogenized water. The regularization method is proposed as one adaptable for the quite limited number of data set in the coastal acoustic tomography.

Keywords: Coastal acoustic tomography, vertical profile inversion, 3-point regularization, Seto Inland Sea
1. INTRODUCTION

The coastal acoustic tomography (CAT) is proposed as an innovative technology to measure the environment variations of coastal and inland seas with strong fishing activity [1]-[6]. The 2D mapping (snap shot) of variable coastal current was the targets of these cat experiments in which the tomography domains are surrounded by multiple acoustic stations and the reciprocal sound transmission was performed among the stations.

The CAT studies have been mainly developed to map the variable structures of depth-averaged current on a 2D (horizontal) plane. However, coastal seas are often characterized by 3D structures such as coastal jets, fronts and upwellings. The variable 3D structures are the final target of CAT to make the simultaneous mapping of complicated environments possible.

Meaningful information on vertical structures can be retrieved when multiple arrival peaks were retrieved in the Kuroshio region [7], [8]. It is worth noting that this vertical profile inversion makes an undoubted step to the 3D mapping in collaboration with the 2D horizontal plane analysis. The purpose of this study is to show the possibility of vertical profile inversion of current and temperature from two arrival peaks data in the coastal sea.

2. SITE AND METHODS

A reciprocal sound transmission experiment was carried out between two acoustic stations, located on both sides of Akinada (HS1 and HS2), located at the central part of the Seto Inland Sea (Fig.1). The station-to-station distance between HS1 and HS2 is 13.769 km after the correct estimate. Notice that this length is determined with the accuracy less than 1m in such a way that the sound speed estimated from the travel time data for the biggest arrival peak is equated to that from the CTD data at the nearest station (station 7) of HPTRI (Hiroshima Prefectural Technology Research Institute). The floor depth is gradually increased from 9 m in HS1 to 56 m in the distance range of 11-13 km from HS1 with the bank of depth 30 m in the distance range of 6-8 km from it, and suddenly slopes up to 10 m in HS2.

Near the sound transmission line HS1-HS2, there is an observation point (station 7) of HPTRI. The CTD data, provided monthly from this organization are used as the validation data for CAT. The range independent ray simulation is also performed by using the sound speed profile, calculated from the CTD data. We can understand how acoustic rays travel in underwater between HS1 and HS2.

The 4 kHz CAT system was set up on the offshore side of breakwater, protecting the fisheries port (Fig.2). The broad-band acoustic transducer of central frequency 4 kHz was suspended down by a rope in water of depth 5 m in front of the breakwater and connected to the CAT system in the plastic box via a cable. The CATS was operated by the 12 V rechargeable battery in connection with the 12V solar panel and the 4kHz sound modified by one period of 11th order M sequence (1.532s) was transmitted every 10 minutes with 24 V power, using two rechargeable batteries and solar panels in a serial connection. The received signals are cross-correlrated with the M sequence replica to increase the signal-to-noise ratio by $20 \log \sqrt{2^{11} - 1} = 33.1$ dB. The first/biggest peak in the correlation waveform is identified as Ray-1 and the second peak corresponding to Ray-2 is found as the biggest peak in the period range of later than 3ms behind the first/biggest peak.
3. INVERSION

The vertical section spanned between two acoustic stations is divided into the horizontal layer of N pieces. The differential travel time ($\Delta \tau_i$) and mean travel time ($\delta \tau_i$) for the i-th ray path travelling between the stations may be expressed in a discrete form by

$$\Delta \tau_i = \tau_i^+ - \tau_i^- = -2 \sum_{j=1}^{N} \frac{l_j v_j}{C_{0j}} \quad (i = 1, 2, \cdots, M) \quad (1)$$

$$\delta \tau_i = \tau_i^+ + \tau_i^- = -2 \sum_{j=1}^{N} \frac{l_j \Delta C_j}{C_{0j}^2} \quad (i = 1, 2, \cdots, M) \quad (2)$$

where $v_j$ and $\Delta C_j$ denote the velocity and sound speed deviation for the j-th layer, respectively. $C_{0j}$ is the reference sound speed for the j-th layer and $l_j$ the arc length of the i-th ray crossing the j-th layer.

The above equations are rewritten in the matrix form

$$y = Ex$$

where

$$y = \{\Delta \tau_i \quad or \quad \delta \tau_i\} \quad E = \left\{ -\frac{2l_j v_j}{C_{0j}} \right\} \quad \{v_i \quad or \quad \Delta C_j\}$$

We shall here apply the inversion with regularization, proposed in geo-tomography, for reconstructing the layered profile of velocity and sound speed deviation [9]. For the upward convex rays, the rays and layers should be numbered from the bottom to surface. The situation is reversed for the downward convex ray. The interrelationship between the
horizontal layers and ray paths is illustrated in Fig.3 for the case of 2 rays and 5 layers (minimum set for the vertical profile inversion).

*Fig.3: Numbering of the ray paths travelling the horizontal layers*

The cost function ($J$) consists of the data misfit and smoothness measure ($H$) of solution vector ($x$):

$$J = (y - Ex)^T (y - Ex) + \lambda x^TH^THx$$

where $\lambda$ is the Lagrange multiplier and the two-order derivative factor $S$ may be expressed by

$$S(x) = \sum_i (x_{i+1} - 2x_i + x_{i-1})^2 = x^TH^THx$$

By minimizing $J$, the expected solution is given by

$$\hat{x} = (E^TE + \lambda H^TH)^{-1} E^Ty$$

where the $\lambda$ is so chosen that the residual defined by $||\hat{x}||^2=||y-E\hat{x}||^2$ is less than a predetermined value
4. RESULTS

Water in this region is well homogenized by the strong tidal current over 1m/s at the spring tide except the warmest season. The weak thermal stratification (1C difference between the surface and bottom) develops only in August and September. One month data during September 2012 are analyzed as a typical example of the stratified season. The range-independent ray simulation is performed by using the CTD data, obtained at station 7 on Sept. 4, 2012 and the result is shown in Fig. 4. Ray-1 is the first arrival ray with the travel time of 9.9898s and travels with the small up-and-down amplitudes around depth 22m. On the other hand, Ray-2 (travel time: 8.9949s) is the bottom bouncing ray with upper turning points ranging from depths 4m to 14m.

![Fig.4: Numbering of depth layers corresponding to the ray paths](image)

The time plots of the path-averaged velocity and temperature are shown in Fig. 5 for the first (red line) and second (green line) arrival peaks. The hourly tidal current is close to 1m/s in the fortnightly period in spite of the worse data quality during 3-11 September. For the period of better data quality (12-30 September), the velocity for the second peak (Ray-2) is a little greater than that for the first peak (Ray-1). The temperature for Ray-1 is rather greater by 0.5C than that for Ray-2.

The time plots of the layered velocity, temperature and sound speed deviation, calculated by the vertical profile inversion for 2-peak and 5-layer are shown in Fig.6. The data are smoothed through the 2-day low-pass filter (LPF) to remove the tidal component from the original data. For the period of better data quality, the upper layer velocity oscillates at 3 days periods, forming the negative mean velocity of about 5cm/s. This oscillation is also visible weakly in the temperature data. The upper-layer temperature is greater by 0.3C than the lower-layer one, forming the maximum temperature on Sept. 19. The sound speed deviation is shifted on the warmer side by 1C for the second half of September due to the reference sound speed fixed on Sept. 4. The 5-layer temperature profiles reconstructed for Sept. 4, Sept. 13 and Sept. 30 are shown in Fig.7 together with a sequence of velocity profiles during the M2 tidal period (12.42 hours) on Sept.14. The temperature profiles for Sept.4 are well compared with the CTD data for September 4 at station 7. The vertical profiles of oscillatory tidal current are roughly reconstructed every one sixth period of the M2 tide. The 3-day oscillation of temperature is also confirmed with the power spectral density diagram for the 5th (shallowest) layer velocity.
Fig. 5: Time plot of path-average velocity and temperature for the first (red line) and second (green line) arrival peaks.

Fig. 6: Time plots of the layered velocity (upper), temperature (middle) and sound speed deviation (lower), calculated by the vertical profile inversion for 2-peak and 5-layer.
5. SUMMARY

The vertical profile inversion is applied to the reciprocal sound transmission data, obtained on September 2012 in Akinada of the Seto Inland Sea, Japan, characterized with strong tidal current and weak thermal stratification. The five-layer profile of velocity and temperature are well reconstructed although the number of data sets is limited to only two. The reconstructed profile of temperature is also roughly validated by the CTD temperature, provided monthly at the nearby station of HPTRI. The usefulness of the present inversion for vertical profile reconstruction is preferable to be further validated in the sea with strong thermal stratification. The reconstructed velocity profile should also be compared with the ADCP data.

6. ACKNOWLEDGEMENTS

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Tomographic mapping of coastal upwelling

generated in Hiroshima Bay, Japan

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Abstract: A coastal acoustic tomography (CAT) experiment of range (4-9) km was performed with four sound transmission and reception stations (acoustic stations), surrounding the northern part of Hiroshima Bay, Japan. The good data set of one-way travel time was acquired along the five transmission lines although significant data lack was produced by a number of oyster rafts distributed widely over the bay. The coastal upwelling generated along the northern shore of Hiroshima Bay by the northerly wind derived from typhoons passing the east side of the bay is the main target of this experiment. The station-to-station distances are corrected to attain the sufficient accuracy of sound speed (temperature) in such a way that the sound speed determined from the travel time data is equated to that calculated by a couple of CTD (conductivity-temperature-depth) data on each transmission line. The inversion (grid method) accompanied by twice the moving average, is applied to reconstruct the horizontal distribution of temperature averaged for the upper 8 m. The initiation, growth and decay processes of coastal upwelling, are mapped with the accuracy of 0.1 °C. The sea level depression of about 0.1m due to the coastal upwelling is also discussed.

Keywords: coastal acoustic tomography, coastal upwelling, inversion, typhoons, sea level depression
1. Introduction

Hiroshima Bay located at the western part of the Seto-Inland Sea is as famous as the biggest oyster aquaculture field in Japan. In the last two decades, the specific sea level rises (SSLRs) over the floor level of the Itsukushima Shrine sometimes occurred at the northern part of Hiroshima Bay [1]. The SSLRs were strongly related to internal seiches and surges, generated by the northerly wind from typhoons which passed the Pacific Ocean off the Kii Peninsula on the east side of the bay. The detailed process of coastal upwelling which triggers internal seiches and surges is another attractive issue to be investigated.

2. Experiment site and method

A coastal acoustic tomography (CAT) [2-6] experiment of range (4-9) km was performed with four sound transmission and reception stations (acoustic stations), surrounding the northern part of Hiroshima Bay, Japan (Fig.1). The 10 kHz sound, phase-modulated by the 11th order M sequence was transmitted every 10 minutes from the broadband transducer (ITC-3013). The experimental region has a relatively flat seafloor ranging from 10 m to 20 m. The station-to-station distances were corrected to measure the sound speed or temperature with a sufficient accuracy in such a way that the sound speed determined from the one-way travel time data was equated to that calculated on each transmission line by a couple of CTD (conductivity-temperature-depth) data after the station positions were corrected through the average of one-month GPS data.

Inside the tomography domain, one temperature array (thermister chain) was deployed from the surface to bottom (0m, 2m, 4m, 7m and 10m) at T1(Fig.1). The CTD casts were executed at ten stations along each CAT transmission line, using the small fishing boat. The CTD data was taken in two days (18th Sep.) after typhoon T1318 was closest to the bay (16th Sep). The CTD data provided not only the oceanographic conditions on 18th Sep., but also the comparison data with CAT.

Fig.1 (a) Location map of the Seto-Inland Sea and the adjacent regions with the trajectories for typhoons T1318 and T1320. (b) Location map of the northern part of Hiroshima Bay with four CAT stations (H1, H2, H3, H5) and ten CTD points (C1-C10). The blue lines connecting the acoustic stations show the sound transmission lines. The tomography domain is divided into nine rectangular sub-domains with red meshes.
For the four-station CAT experiment, the total number of sound transmission lines is quite limited to be six at maximum. The sound transmissions between H1 and H3 were severely interrupted by the oyster aquaculture rafts which existed on the transmission line. Even for the five successful transmission lines, some of sound transmission data were disturbed by the distributed oyster rafts. The travel time data with signal to noise ratios (SNRs) greater than 4 were selected for the following analysis, producing data missing for smaller SNRs. The missing data were reproduced through the linear interpolation of the neighboring data. Notice that the range-averaged current requires the differential travel time data measured with a high clock accuracy while the range-averaged sound speed can be calculated with sufficient accuracy from one-way travel time data. Thus the percentage of missing data is considerably improved by selecting the station with better signal levels. Most important issue in sound speed measurement is to maintain the sufficient accuracy of station-to-station distance for all station pairs.

3. Inversion with five-point moving average

The sound speed deviations averaged along the rays are converted into ones inside the individual sub-domains by the inversion. As for the inversion, there are two kinds of methods: the function expansion method and grid method. The latter method, in which the tomography domain is segmented by the rectangular meshes, is more suitable for the case the data number is so small as in the present experiment. The grid method requires the smoothing process to suppress biases in the expected solution after inversion while the smoothing process is implicitly embedded in the function expansion method.

The inversion problem is here configured as a case of five transmission lines and nine sub-domains. The one-way travel time deviation ($\Delta t_i$) for the i-th transmission line may be expressed by

$$\Delta t_i = t_i - t_{0i} = -\sum_{j=1}^{N} \frac{l_j \delta C_j}{C_0} \quad (i = 1, 2, \ldots, M) \quad (1)$$

Where $\delta C_j$ denotes the sound speed deviation for the j-th sub-domain. $C_0$ is the reference sound speed for the whole domain and $l_j$ the arc length of the i-th ray crossing the j-th sub-domain. The $t_i$ and $t_{0i}$ donate the travel time and reference travel time for the i-th ray, respectively. The equation (1) further reduces

$$y = Ex$$

By applying the singular value decomposition, the transform matrix E becomes

$$E = UA V^T$$

where $U$ and $V$ are the singular vectors spanning the data and solution spaces, respectively and $\Lambda$ is the singular values. The generalized inverse is applied to solve the above equation and the expected solution ($\hat{x}$) is

$$\hat{x} = V\Lambda^{-1}U^T y$$

The smoothed solution ($\tilde{x}$) is obtained through the moving average, based on the sub-domains neighboring to the north-south and east-west and presented with the following weighting matrix $H$.
\[
H = \begin{bmatrix}
1/3 & 1/3 & 0 & 1/3 & 0 & 0 & 0 & 0 & 0 \\
1/4 & 1/4 & 1/4 & 0 & 1/4 & 0 & 0 & 0 & 0 \\
0 & 1/3 & 1/3 & 0 & 0 & 1/3 & 0 & 0 & 0 \\
1/4 & 0 & 0 & 1/4 & 1/4 & 0 & 1/4 & 0 & 0 \\
0 & 1/5 & 0 & 1/5 & 1/5 & 1/5 & 0 & 1/5 & 0 \\
0 & 0 & 1/4 & 0 & 1/4 & 1/4 & 0 & 0 & 1/4 \\
0 & 0 & 0 & 1/3 & 0 & 0 & 1/3 & 1/3 & 0 \\
0 & 0 & 0 & 0 & 1/4 & 0 & 1/4 & 1/4 & 1/4 \\
0 & 0 & 0 & 0 & 0 & 1/3 & 0 & 1/3 & 1/3 \\
\end{bmatrix}
\]

and becomes
\[
\hat{x} = H^T H \hat{x}
\]

4. Results

The temperature array data averaged for sensors at depth range (0-7)m are shown in Fig. 2 together with the sub-tidal (1-day to 1-month time scale) sea level and wind speed from Sep.11 to Oct. 16. The sea level data were corrected by the air pressure at the rate of 1cm/hPa. The strongest northerly wind derived from T1318 blown over the northern Hiroshima Bay on Sep. 16. The northerly wind from T1320 is also visible on Sep. 26, with weaker magnitudes. The 10-cm sea level depression occurred in time of the strongest wind and in a few days later the temperature decreased by 1.1°C for T1318 and 0.5°C for T1320.

The distribution of temperature in two vertical cross sections along the transects C6-C8-C9-C1 and C5-C4-C3-C2 is shown in Fig.3. The 24.1°C water extending up to depth 5m at C1 was exposed to the surface at C6. The surface exposition of 24.1°C water did not exist in the southern transect C5-C4-C3-C2.

The distribution of temperature on the horizontal section of depths 1 m, 3 m and 6 m is shown in Fig.4. Water with temperatures less than 24.0°C is confined to the edge region near the H5 station at depth 1 m. The temperature increased northeastward and reached 24.8°C near the H2 station. The water colder than 24.0°C spread over to the northeast at depth 3m (range 23.9°C-24.4°C) and covered almost the whole region at depth 6 m (range 23.9°C-24.1°C).

The temperature variations, determined by the inversion of travel time data are shown in Fig. 5 with the contour plots at 6-hour interval after the maximum northerly wind (4:00 16th Sep.) due to T1318. The temperature started to decrease from the southern half of the tomography domain at 04:00 17th Sep and became smaller than 23.8°C in most regions except the northern corner.
Fig.2 Time plots of the upper 7m temperature and the sea level and wind speed for the sub-tidal range from Sep. 11 to Oct. 16. The vertical down-arrows point out the occurrence of the maximum northerly wind, caused by typhoons T1318 and T1320.

Fig.3 Contour plots of temperature in the vertical cross sections, drawn from the CTD data on Sep. 18. The left and right panels are for the transects C6-C8-C9-C1 and C5-C4-C3-C2, respectively.

Fig.4 Contour plots of temperature on the horizontal sections, drawn from the CTD data on Sep. 18. The left, middle and right panels are for the depth of 1m, 3m and 6m, respectively.

Fig.5 Contour plots of the temperature distributions on the horizontal section, reconstructed by the inversion after the passage of T1318 at 4:00 16th Sep.

5. Summary and Discussion
The horizontal distributions of temperature, reconstructed by the inverse analysis of the CAT data are compared with those constructed from the CTD data (Figs.6a and 6b). The CAT data ranging from 24.1°C to 24.35°C is in good agreement with the CTD data ranging from 24.05°C to 24.35°C. The warmer water over 24.3°C exists in the northeastern part of the domain on both cases of CAT and CTD. The difference of temperature between both the data is in the range of (-0.1~0)°C in most regions (Fig.6c). The correctness of the inversion analysis, based on the grid method accompanied by the moving average, is well validated by the above comparison.
The sea level depression of (0.05–0.1) m was generated along the northern shore of Hiroshima Bay by the northerly wind, caused by T1318 and T1320. Water around the northern shore was colder by about 1°C in a few days after the sea level depression. This decrease of water temperature was also found in the inversion result of the CAT data. It is likely that the colder water is produced by the coastal upwelling which occurs along the northern shore of Hiroshima Bay. Further study is required to elucidate the dynamic dependency of the sea level depression and the appearance of colder water. The transition process from the coastal upwelling to internal seiches or surges may be interrupted by a sequential attack of typhoons in this study.

Fig.6 Comparison of temperature distributions for the upper 8 m, obtained on the horizontal section by the (a)CAT and (b)CTD experiments on 18th Sep. The temperature difference between (a) and (b) is also shown in (c).

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Session 5

Acoustics in Polar Environments

Organizers: Jaroslaw Tegowski and Alexander Gavrilov
MEASUREMENTS OF THE NOISE FIELD DIRECTIONALITY IN AN ARCTIC, GLACIAL FJORD

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Abstract: Ocean-ice boundaries in the Arctic are dynamic regions that are receiving increasing attention as sensitive indicators of shifts in climate. Recently, there has been interest in using the underwater ambient noise in the fjords of marine-terminating glaciers as a new tool to study glacier dynamics. Although there already exist many useful measurement systems for studying glaciers, the use of underwater sound presents some attractive features: sound can propagate long distances underwater and equipment to monitor underwater ambient noise for a year or more is available and relatively inexpensive. Monitoring glacier dynamics using underwater ambient noise assumes a knowledge of the mechanisms producing the noise and the relationship of those mechanisms to their source spectrum and generation statistics. Here are presented measurements of the underwater ambient noise made close to Hans Glacier in Hornsund fjord, Svalbard during the summer of 2013. The objective of the field study was to see if low frequency (100 Hz – 3000 Hz) and high frequency (2000 Hz – 5000 Hz) noise within the fjord could be associated with distinct and separate sources. The measurement system consisted of two broadband hydrophones separated by 43 cm along a horizontal axis and oriented with a magnetic compass and tilt sensor. This arrangement enabled the creation of maps of the ambient noise directionality at different locations within the bay containing Hans Glacier, and tested the idea that different physical mechanisms are associated with different frequency bands. Melting ice, ice-wave interactions and glacier calving events were all observed to contribute to the underwater noise field and the ambient noise field directionality was found to be a strong function of frequency.

Keywords: Arctic, ambient noise, noise field directionality, glacier
1. INTRODUCTION

There is a long history of underwater ambient noise measurements in the Arctic, dating back to the 1960’s [1-3]. Much of the early work focused on the statistical properties of noise associated with sea ice and its propagation through the ocean. More recently, attention has turned to undersea noise in the fjords of marine-terminating glaciers in Alaska, Svalbard and Greenland[4-7]. Motivated by a desire to understand the increasingly important role global warming plays in the Arctic[8], these studies are designed to exploit the properties of underwater noise to monitor and study tidewater glacier dynamics. Studies of glaciers based on underwater ambient noise can take advantage of the fact that measurements are relatively easy to make using inexpensive instruments over long (order 12 month) time periods.

The use of ambient noise to study the ocean and noise generation mechanisms is generally called Ambient Noise Oceanography (e.g. [9]), and its successful application requires the physics of propagation in the ocean and the spectrum of noise sources to be understood. The current challenge facing those who wish to exploit undersea noise to study glaciers is to form quantitative links between noise level and process of interest, such as calving rates, ice melt rates and freshwater outflows. This is a non-trivial problem and its resolution will require careful noise measurements coupled with extensive, non-acoustic characterization of these glacial processes. Here we make the case that characterizing the noise field directionality will provide a valuable tool for solving the forward problem.

The present study represents a characterization of the ambient noise field in a tidewater glacial fjord in terms of its frequency-dependent directionality in the horizontal. The motivation for the study is a need to substantiate the working hypothesis that distinct glacial processes, such as calving and ice melting, radiate noise in distinct spectral bands and also create a frequency-dependent directionality to the noise field, which is determined primarily by the distribution of noise sources throughout the fjord.

In July 2013, we participated in the polar expedition to Hornsund fjord and used the facilities of the Polish Polar Research Station Hornsund Stanislaw Siedlecki named in Isbjørnhamna bay, owned and operated by the Institute of Geophysics, Department of Polar and Marine Research, Polish Academy of Sciences, to support the field study. The general study site was Hornsund fjord, and the data presented here were taken in Isbjørnhamna, in the fjord just south of the terminus of Hans glacier. Analyses of sounds previously recorded in the Brepollen area of eastern Hornsund show very high levels of underwater noise, despite the fact that during measurements, the weather was calm, with no wind, rain or breaking waves, and the fjord was empty of vessels[4,6]. The present study site location is shown in Fig. 1a and a photograph of a drifting iceberg can be seen in Fig 1b. Ice floes were evident on most days during the month of July and were particularly numerous after a storm. The dominant tidal circulation in the fjord was counter-clockwise, and calving icebergs typically flowed along (or shoaled and grounded on) the northern shore of Hornsund fjord on their passage from the glacier to the open ocean.
Fig.1. a) The location of Hans Glacier (77°05’N, 15°38’E) in Hornsund fjord, b) an iceberg drifting in front of Hans glacier. The 30 m – 40 m tall terminus of the glacier can be seen in the background. The iceberg in the foreground projects ~10 m from the water.

2. MEASUREMENT METHODOLOGY

Eight sites were surveyed over 3 days for underwater ambient noise between 250 – 1000 m south of the terminus of Hans glacier. Figure 2 shows the location of the measurement sites, marked with a white + symbol and superposed on a satellite image of the glacier (AlosAvnir 2009.07.11). The white lines projecting from the site markers show the track of the drifting boat in those cases when the boat was not anchored. The red lines show bathymetric contour lines in the fjord, with depth annotated in yellow. The inset at the bottom of the figure is discussed in the results section.

Noise was recorded for 30 – 60 minutes at each site using the Directional Acoustic Buoy (DAB) system. DAB consists of two International Transducer Corporation 6050C general-purpose hydrophones connected to a Sony TCD-D8 digital audio tape recorder. The ITC6050C hydrophone has a sensitivity of -162 dB re 1Vrms re 1μPa and a usable bandwidth in excess of 60 kHz which, when combined with the Sony recorder, provided an upper frequency limit of 22 kHz. Each segment of ambient noise was preceded by approximately 60 s of white noise of known level connected to the Sony recorder to provide a calibration signal for the subsequent data analysis. The two hydrophones were mounted 0.43 m apart on a horizontal bar and connected to a surface float by a 1 m long vertical support. A navigation system with an electronic magnetic compass and two tilt sensors housed in a splash-proof pelican case were attached to the surface float and provided three bearing and tilt measurements a second. The compass worked well under the most benign ocean conditions, but the strong magnetic inclination (82°) at Svalbard’s latitude caused problems for the instrument if its tilt from the horizontal exceeded approximately 5°. For this reason, DAB was modified part way through the deployment by adding crossed spars and a ships compass to the top to provide control and visual confirmation of its heading. Compass, tilt and a GPS position fix were logged to a
computer simultaneously with the acoustic data. Deployments were staged from a rubber boat which was left to freely drift or was anchored, depending on the deployment site. Registering noise closer than roughly 300 m from the glacier terminus is hazardous as ice calving events are accompanied by the generation of tsunamis, which can overturn small boats.

Noise data was transferred from audio tape to a computer using a dedicated digital transfer device. These data were then converted to maps of noise directionality using the following signal analysis procedure. Since we have only two hydrophones, it is not possible to determine the noise field directionality at many separate frequencies. However, it is possible to filter the noise time series into a small number of distinct bands (in this case, 2) and then compute the dominant arrival angle of the noise within that band using a time delay analysis. This simple process is effective, but cannot resolve the broadside ambiguity in directionality inherent in a single-axis hydrophone array. That is, one can determine the propagation direction of waves with a component of wave front normal to the array axis, but not when the wave fronts are parallel to the array axis. To resolve this ambiguity, the array was physically rotated through 90° every minute or so, enabling the direction of noise sources persistent from more than a few minutes to be determined unambiguously. This mitigation procedure solved the problem with varying degrees of success through the deployment. The sources of low frequency noise were observed to be stable over many minutes (with the exception of ice calving events), and the array ambiguity was resolved. Noise sources in the high frequency band were more variable, and it was not always possible to process the ambiguity.

Maps of noise field directionality have been created for energy in two spectral bands: 100 Hz – 3 kHz (the low frequency band) and 2 kHz – 5 kHz (the high frequency band), resulting in an overlap in the bands between 2 kHz – 3 kHz. The lower frequency limit of 100 Hz was set by the data acquisition equipment. After filtering, 20 min – 30 min recordings of data were divided into shorter segments and subjected to a cross-correlation analysis to determine the time delay in signal arrival between the two hydrophones (the delay time). The low frequency band required the analysis of more data than the high frequency band to arrive at a stable estimate of delay time: segments lengths of 1.36 s and 0.34 s were chosen for the low and high frequency bands respectively. Once determined, the delay time was converted to arrival angle by taking the inverse cosine function of the measured delay of the highest peak in the cross-correlation function, normalized by the maximum possible delay for a wave front propagating along the array axis.

3. RESULTS

Results from the analysis of a single period of recorded noise from site 3C are shown as an insert at the bottom of Fig. 2. The analysis described in section 2 above was applied to the data segment. The magnetic compass deployed in the DAB buoy allowed the conversion of the arrival angles to compass headings, which were then used to generate a rose plot of dominant arrival angle in the low and high spectral bands. The rose plot at the bottom of Fig. 2 presents a histogram of frequency of occurrence of arrival angle arranged onto a rose compass, with high and low frequency spectral bands respectively coloured blue and red. It is clear from the plot that the noise field directionality is quite different in
the two spectral bands. Moreover, arrivals tend to come from a relatively narrow band of angles spread around a few dominant directions.

![Diagram of measurement sites and bathymetry](image)

**Fig. 2.** The location of the measurement sites, marked with white + symbols. Bathymetry contour lines are red and annotated with depths in yellow. The inset at the bottom of the figure shows a rose plot of dominant arrival angle for noise recorded at site 3C in two spectral noise bands (see section 3 for further details).

### 4. CONCLUSIONS

We have measured the horizontal directivity of ambient noise around the terminus of a tidewater glacier in Hornsund fjord during the summer months using a 2-hydrophone system nicknamed DAB (Directional Acoustic Buoy). These measurements were made to test the hypothesis that ambient noise in different spectral bands are associated with distinct physical mechanisms, and are therefore expected to exhibit distinct patterns of arrival for a given measurement location. The data strongly support this hypothesis. In addition, the analysis demonstrates that during the presented measurement interval, the noise field horizontal directionality was dominated by arrivals from a few, well-defined directions. The implication is that the noise field was generated by a few single sources (such as a fresh water outflow region in the low frequency band) or clustered groups of sources (such as melting icebergs in the high frequency band), which resulted in the well-defined pattern of arrivals observed. These data demonstrate the advantages that can be gained by deploying recording systems with multiple hydrophones.
The next challenge in this line of research is to quantitatively relate the location and intensity of underwater sound sources around the glacier terminus and throughout the fjord to the measured noise intensity and horizontal directionality. This will require characterizing the propagation environment in the fjord, which is complicated by the presence of surface layers of low salinity melt water, and a greater understanding of the spectral properties and intensities of noise sources than is currently available.

5. ACKNOWLEDGEMENTS

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ZOOPLANKTON DISTRIBUTION STUDIES COMBINING ACOUSTICAL AND OPTICAL OBSERVATIONS

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Abstract: Concurrent acoustical and optical measurements have a great potential to describe zooplankton assemblages over large temporal and spatial scales. It is difficult to assess thorough information on zooplankton distribution with traditional methods (e.g. nets), that provides only discrete and sparse information on biomass and community structure of zooplankton, therefore use of alternative methods, should be taken into consideration. Acoustic echosounding allows fast, non-intrusive, and relatively cheap environmental studies with high temporal and spatial resolution. Laser Optical Plankton Counter (LOPC) proved to be well suited to multi-scale studies of zooplankton communities. LOPC delivers real-time information on zooplankton size spectra and abundance. Presented results are based on the data collected on the West Spitsbergen Shelf in the summer seasons of 2012 and 2013. The high frequency 420 kHz acoustics was supplemented by LOPC measurements along the transects, additionally net sampling, that provides information on zooplankton taxonomy, was taken at fixed points. Size spectra measured by LOPC were used as input parameters in “high-pass” model of sound scattering on fluid-like particles. Model output values of acoustic backscattering strength were compared to values obtained by echosounding. In most cases there is a good agreement between measured and modeled values, except conditions of very low zooplankton abundance and events of fish presence. Zooplankton size structure measured by LOPC is helpful in validating and refinement of “high-pass” acoustic model for specific set of scatterers. This gives a possibility to determine the theoretical backscattering strength of zooplankton aggregations. Implementing two complementary methods allows to obtain fast and thorough information on zooplankton patches and fills the gap in comprehensive studies of the Arctic zooplankton in the frontal zone.

Keywords: high-frequency acoustics, Arctic zooplankton, LOPC, sound scattering model
1. INTRODUCTION

Research focused on zooplankton, a key component of the ecosystem linking primary producers with higher trophic levels. Zooplankton community is characterized by sparse distribution throughout water column with patches caused by either environmental or behavioral factors [1]. It is difficult to assess thorough information on zooplankton distribution with traditional methods (e.g. nets), therefore use of alternative methods, should be taken into consideration. Acoustic sampling methods allow fast, non-intrusive, and relatively cheap environmental studies with high resolution [2]. Moreover, optical instrument for assessing zooplankton abundance has proved to be well suited to studies of zooplankton communities [3, 4].

Continuous echosounding was conducted with transducer working at a frequency of 420 kHz, giving a 2-dimensional backscatter field in the water column along the ship route. Concurrently with high frequency acoustical measurements, complementary methods were used – Laser Optical Plankton Counter (LOPC) measurements delivered high resolution information on zooplankton size spectra and abundance, while sampling nets provided information on zooplankton taxonomy.

Information on population size spectra allowed implementation of the mathematical model of acoustic scattering on the Arctic zooplankton community. The last part of research was a comparison between measured values of backscattered acoustic energy and model-calculated results, with use of the zooplankton size spectra obtained by LOPC.

Presented results were obtained in the fjords of West Spitsbergen, investigated during research cruise in the summer of 2013.

2. STUDY AREA

Zooplankton studies were carried out in the two West Spitsbergen fjords, Hornsund and Kongsfjorden, during cruise of r/v Oceania in the summer of 2013, as a part of the GAME (Growing of the Arctic Marine Ecosystem) project. The Hornsund fjord is regarded as cold, under the influence of South Cape Current, while Kongsfjorden is influenced by the warmer West Spitsbergen Current, that carries Atlantic waters.

![Fig. 1 Area of investigation – West Spitsbergen fjords: Kongsfjorden (north) and Hornsund (south)](image_url)
3. MATERIALS AND METHODS

Acoustical measurements were supplemented by sampling nets and LOPC. A comparison of different methods is required to assess more thorough information, as each sampling tool suffers its own inefficiencies.

Acoustic measurements were done with a DT-X echosounder (BioSonics Inc., Seattle, USA), working at a frequency of 420 kHz. Downward-looking acoustic transducer was mounted on the ship, 1 meter below water surface, by a special frame. The construction allowed the ship to proceed with speed of approximately 3 knots. Pulse length was set at 0.3 ms and trigger value at 2 Hz. The output of the echosounder is a volume backscattering strength $S_V$, which is a logarithmic measure of the volume backscattering coefficient $s_v$, being the sum of the backscattering cross-sections of all scatterers enclosed in the ensonified unit water volume. Output $S_V$ values were averaged over 1 meter depth layers and 5 second time intervals. The frequency of 420 kHz gives an acoustic signal wavelength of about 3 mm, thus it is capable of detecting individual zooplankters with the equivalent spherical radius of 0.5 mm. It is based on the criterion of detectability: $2\pi \text{ radius/wavelength} > 1$ [5].

The zooplankton in situ samples were additionally collected with WP2 sampling net (with 500 $\mu$m mesh size) on chosen stations. Samples were preserved and returned to the IO PAS Marine Ecology Department laboratory for microscopic analysis, where they would be identified, counted and measured.

Concurrently with echosounding, Laser Optical Plankton Counter - LOPC (Brook Ocean Technology Dartmouth, Canada) was hauled along the transects. LOPC is the in situ equipment which provides abundance and community size structure of plankton and particles in water environment. The technical specifications allow to obtain particles size spectra in range of 100 $\mu$m to 35 mm Equivalent Spherical Diameter (ESD). LOPC was working in an undulating mode through the water column from the surface to maximum 50 meters depth, pulse rate was established to 2 Hz.

Quantitative relationship between acoustic backscattered signal and zooplankton abundance, its size and taxonomy structure is a complicated problem. The acoustic scattering intensity depends on wave frequency and scatterers’ size, shape, orientation, age, concentration and material properties (sound speed and density ratios). Because of its complexity, evolution of acoustic models is a great challenge. Many different models, which vary in accuracy and generality, have been developed within the past decades. First scattering models treated zooplankton as a homogeneous fluid sphere [6, 7, 8]. More sophisticated models have been introduced taking shape and material properties of animals [9, 10, 11] into account. In this study we used so called “high-pass” model for a fluid sphere introduced by Stanton [9], which proved to be a good approximation in the Arctic waters.

Zooplankton size spectra obtained by LOPC were used as input parameters for theoretical acoustic backscattering model of zooplankton. To validate implemented model, the sum of the modeled values of backscattering cross-sections of all scatterers were transformed into total $S_V$ and compared with real $S_V$ values measured by echosounder in the water layer of LOPC towing depth. For comparison, acoustic data were averaged over 1 meter depth layers and 10 transmissions (5 s). The same applied to optical data which were averaged over the same time and depth intervals. The zooplankton sound speed contrast was established at $h=1.027$ and density contrast at $g=1.0$ for the Arctic copepoda species Calanus finmarchicus [12]. The collected data consist of 13 hours of concurrent acoustical and optical sets of measurements, 5 hours in Hornsund and 8 hours in Kongsfjorden. The resultant output was divided into 30 minutes long data series.
4. RESULTS

From the set of collected data, we present one exemplary file taken in Kongsfjorden on 06 August 2013 18:49 UTC. LOPC towing route was plotted on the acoustic echogram image (Fig. 2). For each time dependent towing depth of LOPC the corresponding values of acoustic backscattering strength were chosen (red curve) and compared with the sum of the modeled values (blue curve) (Fig. 3). The repetitive maxima and minima reflect changes in $S_V$ values, caused by undulating towing mode of LOPC with depth spanning from 0 to 25 meters. The upper layer is more abundant in the zooplankton which is reflected in relatively higher $S_V$ values. The visible peak of measured $S_V$ value is probably caused by fish presence, which could not be detected by LOPC. The linear correlation coefficient between measured and modeled values of $S_V$ for the whole 30 minutes long data set and entire studied water column equaled to $R = 0.72$ (Fig. 4).

![Exemplary echogram with LOPC position during undulating mode](image_url)
Fig. 3 A comparison of measured and modeled values of backscattering strength averaged over 1 m depth layer and 10 transmissions

Fig. 4 Correlation of measured and modeled values of backscattering strength

5. DISCUSSION

The idea of this research was to conduct synchronous acoustical and optical measurements and compare them. Concurrent acoustical and optical measurements have a great potential to describe zooplankton assemblages over large temporal and spatial scales. The echosounding supported by LOPC working in an undulating mode through the water column gives opportunity to investigate different water masses and to obtain zooplankton abundance in the chosen water layers. Moreover, LOPC is capable of assessing rapid and continuous characterization of zooplankton distribution concurrently with environmental parameters. The comparison of optical and acoustical results was performed. In most cases there is a good
agreement between measured and modeled values of backscattering strength $S_V$, except conditions of very low zooplankton abundance and events of fish presence.

6. ACKNOWLEDGEMENTS

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AN ACOUSTICAL STUDY OF GAS BUBBLES ESCAPING FROM MELTING GROWLERS

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\textbf{Abstract:} According to the most recent studies, a steady and consistent increase in the level of underwater noise is observed in Arctic fjords. This phenomenon is mainly caused by intense melting of tidewater glaciers and other associated effects. An important component of the underwater sound budget is the noise generated by gas bubbles escaping from melting icebergs and growlers. In July 2013 we carried out a comprehensive study of the Hans glacier in Hornsund fjord, Spitsbergen with logistic support provided by the Polish Polar Station located in Ishjornhamna. One of the main objectives of the study was to make synchronous video and sound recordings of gas bubbles released by melting growlers. The research was conducted in the shore area and also in a small tank, in which sounds were registered using an HTI-96 hydrophone and WildLife Acoustics SM2+ recorder. The hydrophone was placed a few centimetres from blocks of ice in order to record sounds generated by erupting single bubbles or chains of bubbles. A synchronized video documentation was made from a distance of several centimetres. Analysis of both datasets allowed us to determine the link between the type of bubbles released (e.g. single, chains, several parallel, different sizes, etc.) and the emission of sounds. Analyses of transient events were conducted using a wavelet technique and spectral analysis using both the full spectrum and 1/3-octave frequency bands. The potential mechanisms of sound generation are discussed.

\textbf{Keywords:} Arctic, ambient noise, tank and field experiments, ice, growlers
1. INTRODUCTION

Global warming of the Earth’s surface is particularly evident in the Arctic, where the ice cover is melting at a rapid pace [1]. This fact is reflected in the high level of ambient noise observed in Svalbard fjords with marine-terminating glaciers [2,3]. Starting in 2009, we have conducted regular acoustic measurements of this phenomenon, in order to estimate the level(s) and nature(s) of the underwater ambient noise as an indicator of climate change. Analysis of the first measurements show that ambient noise level and its spectral, wavelet and statistical features are significantly different between the Hornsund fjord, which is surrounded by calving glaciers and the Murchison fjord, which is covered by marine ice floes [2,3]. Analyses of sounds recorded in the Brepollen area of Hornsund show very high levels of underwater noise, despite the fact that during measurements, the weather was calm, with no wind, rain or breaking waves, and the fjord was empty of vessels.

Considering the lack of other loud noise sources, we concluded that physical processes accompanying melting glaciers are the primary source of the high levels of underwater sound observed in Hornsund fjord. We found that, for low frequencies (below 300 Hz), the dominant sources of sound are cracking and calving glaciers. But for measurements made in different parts of the Brepollen area, the maxima of spectra range between 1 kHz and 2 kHz. We assumed that the main source of these sounds are the effects associated with the bursting air bubbles released from the melting growlers, released from the glaciers. To better clarify this effect, in 2012, we measured the melting of small growlers in the anechoic tank of Gdansk Technical University, Poland. The ice for the experiment was collected in the vicinity of the Hans glacier in Hornsund (Fig.1.a) and transported in the deep freezer of R/V Horizont II back to Poland. The sounds of melting growlers were recorded using three hydrophones in parallel [4].

In July 2013, we took part in the polar expedition to Hornsund Fjord using the facilities of the Polish Polar Station in the Isbjørnhamna bay. In addition to field measurements of ambient noise directivity, synchronised air video and underwater sound recordings of calving events and the dynamics of water masses, we recorded the sound

![Fig.1. a) Hans Glacier location (77°05’N, 15°38’E) in the Hornsund fjord, b) example of growlers produced by calving Hans Glacier (visible behind the Baranowski peninsula) - photo taken from the shore of the Polish Polar Station in Isbjørnhamna.](Fig1.png)
radiated by gas bubbles escaping from growlers both in the coastal zone of the fjord and in a small tank, away from other potential sound sources.

2. MEASUREMENT METHODOLOGY

The first field measurements were taken in the coastal zone of Isbjørnhamna, which was full of growlers drifting from the calving Hans Glacier (Fig.1.b). The ambient noise was recorded by one of us dressed in a diving suit, who kept a hydrophone close to growlers in the shallow water area. During the measurement, the weather was calm, with no wind, rain or breaking waves in surf zone and the fjord was empty of vessels. Such particularly “clean” experimental conditions provided noise recordings dominated by the melting of glaciers in the Hornsund fjord and surrounding the measurement site. All measurements were made using an omnidirectional broadband HTI-96 hydrophone, deployed at a depth of ~0.5 m and connected to an SM2+ WildLife Acoustics recorder (sampling at 96 kS/s and with 16-bit resolution).

The controlled, laboratory experiments were conducted in a small tank, which contained selected pieces of growler floating in the water (Fig.2.a). A waterproof torch was placed at the bottom of tank and directed upward to provide illumination of the melting ice. The HTI-96 hydrophone was placed a few centimetres from the illuminated face of the melting ice. The close proximity of the hydrophone to bubbles released from the ice face enabled unambiguous association of bubble release with noise emission. Synchronised with the SM2+ recorder, a high-speed camera with its lens 20-30 cm from the piece of ice added video information.

![Fig.2. a) Experiment setup, b) Close-up view of a growler filled with gas bubbles (sizes ca. 1-3 mm).](image)

A typical chunk of ice (a diameter of ~20 cm) formed from the division of growlers have been selected for the experiment. The air bubbles trapped in the ice used in the experiment are visible in Fig.2.b.

3. RESULTS

Our previous observations show that the dominant frequency band of noise recorded in a fjord containing significant quantities of melting ice is from 1 kHz to 2 kHz [2]. Also the
spectrum of a 10-minute record of ambient noise in the coastal waters of Isbjørnhamna bay (Fig.3) has peaks in the range from 1 kHz to 2 kHz, corresponding to resonant frequencies of bubbles of diameter in the range of 1.6 mm to 3.2 mm. The maximal noise spectrum level in this range is 97 dB re 1\mu Pa^2/Hz. Apart from low-frequency ambient

![Fig.3. Smoothed spectrum of ambient noise recorded in a field of melting growlers (Fig.1.b).](image)

noise (< 300Hz), originating mainly from ice collisions with each other [4], the frequencies higher than 1 kHz mainly result from bubble release from the ice.

Analysis of the concurrent video and sound recordings of ice melting in the small tank show a few specific types of air bubbles escaping from the melting ice. We did observe the release of single bubbles of different sizes, but the release of chains of small bubbles from a single cavity in the ice face was the most symptomatic. An example of chain

![Fig.4. Consecutive movie frames of chain of bubbles escaping from melting ice.](image)
release is presented in Fig. 4, which shows 16 consecutive high-speed frames of a typical bubble chain. In Fig. 4 The ice wall from which the air bubbles can escape is visible on the left of each frame, and part of the measuring hydrophone on the bottom right. The first small bubble appears at bottom left (frame 1). In frames 2, 3 and 4, a chain of small bubbles escaping from the ice wall is visible, as it flows up to the water surface (next frames).

Figure 5.a shows a 0.3-s snapshot of pressure recorded for the frames of Fig.4. Air pressure in the cavities can be up 5 atmospheres [5]. During the release of bubbles from ice, the pressure decreases, this is reflected in the increase in distance between successive peaks and a decrease in their amplitude.

Figure 5.b shows the corresponding spectrogram. The escape of bubbles generates sounds across a large range of frequencies, up to 48 kHz here (the highest achievable limit in this recording, sampling at 96 kS/s) and decreases with the lowering of gas pressure in the cavity. In Figure 5.c, a continuous wavelet transform of the pressure using Morlet wavelets is shown. The coefficients of the transformation precisely indicate jumps in the acoustic pressure and increasing distance between consecutive pressure peaks.

4. CONCLUSIONS

The shape of air bubbles trapped in glaciers is mostly tubular [5], because they are formed as the result of compression of snow filled with air. The previous interpretation of
sound generation by bubbles assumed that, after bursting of the thin ice wall, the escaping air forming the bubbles would oscillate with the Minnaert frequency $f$ [6]:

$$f = \frac{1}{2 \pi} \frac{3 \gamma p}{R \rho},$$  \hspace{1cm} (4.1)

where $R$ is the radius of the bubble, $\gamma$ is the ratio of heat capacity of the gas inside the bubble at constant pressure to that at constant volume ($\gamma = 1.403$ for air close to 0°C [23]), $p$ is the hydrostatic pressure (1 atm = 101,325 Pa) and $\rho$ is the density of water (1,000 kg/m³). The frequency peaks visible in spectrum (Fig. 3) correspond to bubbles of 1.6 to 3.2 mm. But the very high frequencies observed cannot be explained solely as the natural breathing mode oscillations of bubbles released from the ice. The analyses of sound generation should include effects of resonant vibrations of sound inside the ice cavity and whistling effects at its edge. These effects are possible sources of the high frequency sounds observed in the tank experiments. The proper interpretation of high-frequency radiation associated with bubble release demands more precise experiments and modelling of sound generation by bubbles escaping from the ice, both under progress now.

5. ACKNOWLEDGEMENTS

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REFERENCES

THE SOUNDSCAPE OF THE FRAM STRAIT MARGINAL ICE ZONE

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Abstract: A series of acoustic experiments were conducted in the Marginal Ice Zone (MIZ) of the Fram Strait in the years 2010-12 under the Waves-in-ice Forecasting for Arctic Operators (WIFAR) project led by NERSC. The focus of this paper is results from ambient noise measurements with fields of sonobuoys deployed in the MIZ from the open ocean to compact ice under varying environmental conditions. Noise spectra (10 Hz – 1 kHz) are presented and categorized by environmental parameters that include sea state, wind force and direction, ice concentration, and ocean swell. Scatter plot representations of noise data are explored as a tool to infer local ice conditions. The noise fields also included components due to marine mammals and distant seismic exploration.

Keywords: Arctic acoustics, ambient noise, Fram Strait
1. INTRODUCTION

One of the most dangerous places for operators in the Arctic is in the Marginal Ice Zone (MIZ) when ocean waves of high amplitudes propagate into the MIZ from the open ocean. It is therefore important to be able to predict situations so that the operators can make their precautions. This is the focus of the ongoing project WIFAR: Waves-in-Ice Forecasting for Arctic Operators [1]. WIFAR covers wave-in-ice modeling, integration of waves-in-ice into ocean forecast models and observation of waves in the MIZ using passive acoustics and accelerometers. The response to waves in ice is break-up of ice floes and floe collisions caused by compression and decompression of the ice field due to wind, wave and ocean circulation. These responses can be observed by accelerometers and tilt meters drifting with ice floes, or by hydrophones suspended beneath the ice floe or integrated into sub-surface moorings. Hydrophones in moorings is the only long-term monitoring possibility in the MIZ. The challenge is to obtain data in near real time, either via acoustic communication with moorings in open water, or via surface connection or cables to shore, or via installations with satellite communication.

Two kinds of passive acoustic experiments were conducted in the Fram Strait under the WIFAR project. A drifting ice station equipped with a hydrophone array recorded ambient noise continuously over several days, for correlation with data from accelerometers and a weather station. The second kind of experiments used sonobuoys deployed from a P-3C aircraft in different ice and environmental conditions in the MIZ. The aim of these experiments was to study the spatial variability of ambient noise over a large area. This paper discusses the use of ambient noise measurements (from the second set of experiments) to infer information about environmental conditions in the MIZ. Ambient noise measurements also have relevance to monitoring of the habitat of marine mammals.

2. THE CHARACTERISTICS OF THE MARGINAL ICE ZONE

It is well established that low-frequency natural sound in the open ocean is related to sea state. In the ice edge zone the primary sound-generating mechanisms are due to long ocean waves (swell) propagating into the ice pack causing increased internal ice stress and break-up of ice floes [2]. The MIZ comprises highly variable ice conditions ranging from open water to ice-covered regions composed by a mixture of new frozen, first year ice and multi-year ice. During on-ice wind and wave conditions a compact ice region is formed, and the combination of compression of the ice field and waves propagating into the ice pack creates several sound generating mechanisms. The MIZ is a noisy environment with relatively large variability in noise levels [2]. During on-ice wind/wave conditions, low levels of ambient noise is found deeper into the ice pack where the floes are larger and waves have been attenuated such that they no longer cause sound generating responses in the ice. During off-ice wind conditions or low winds, noise from ice and breaking waves in the open ocean is low and the ambient noise level is low. During such conditions, the variations in noise levels in the entire MIZ are lower, but a reduction in noise levels with distance from ice edge has been observed. The most significant minimum in sound levels in the outer part of the MIZ is in areas covered with grease ice. Large areas of grease ice form under cold and calm conditions, and remove all noise generating mechanisms at the surface [2]. Different ice conditions can be seen in Figure 1.
Marine mammals are frequently present in the MIZ of the Fram Strait. Analysis of a year-long recording from a moored array passive listening system at 79° N in the western part of the Fram Strait showed that seasonal variability in vocalization varies between species, e.g., bow head whale calls are heard year-round and blue whale and fin whale calls are heard parts of the year [3]. Most of the year, signals from seismic airguns can dominate the low frequency portion of the Fram Strait soundscape [3].

As soon as the natural, mammal or human generated sound propagates from its origin it will be flavoured by the oceanography and ice conditions. Below ice-covered regions an oceanographic layer with cold and less saline water causes channelling of acoustic energy; this layer diminishes in thickness across the MIZ and disappears in the open ocean. Frequency dependent attenuation of the acoustic signal will be introduced due to reflection and scattering from the rough underside and edges of the sea ice. The acoustic propagation conditions in the MIZ of the Fram Strait were studied extensively in the 1980’s [4], and more recently in the EU funded project ACOBAR [5] and in the ongoing UNDER-ICE project [6] funded by the Research Council of Norway. Propagation conditions are important to acoustic communication and navigation, and for acoustic tomography systems in travel-time inversion for ocean temperature structure [7,8].

3. MEASUREMENTS AND DATA PROCESSING

Figure 2 shows a map of the Fram Strait, the deep water area between Spitsbergen and East Greenland. Deployed acoustic and oceanographic equipment in this area include a tomography triangle with four acoustic moorings (green and light blue), an oceanographic mooring across the strait (yellow) operated under the ACOBAR project [5], and RAFOS sources to aid glider navigation (red). The black box indicates the deployment area for sonobuoy fields used to record ambient noise under the WIFAR project. The area is close to an area of similar measurements during the MIZEX campaigns in 1985 and 1987 [2]. Three missions were flown with P-3C aircraft of the Royal Norwegian Air Force in October 2010, June 2011, and March 2012, respectively. On each mission, approximately 20 sonobuoys were dropped within the area (size 150 km by 150 km) along predefined patterns that included a series of buoys along 0° W from open water to compact ice, a series of buoys approximately E-W along the direction of ocean swell, and additional buoys within the ice zone. Supporting environmental information included ice and ocean
observations from the aircraft, ocean wave and wind information from the Norwegian Meteorological Institute (includes the WAM ocean wave forecast model), and RadarSat images over the experiment area. Table 1 summarizes details of the experiments and the environmental conditions.

![Map of the Fram Strait experiment area, with acoustic and oceanographic moorings. The black box indicates the area of sonobuoy deployments 2010-12.](image)

**Fig. 2:** Map of the Fram Strait experiment area, with acoustic and oceanographic moorings. The black box indicates the area of sonobuoy deployments 2010-12.

<table>
<thead>
<tr>
<th>Experiments</th>
<th>WIFAR October 2010</th>
<th>WIFAR June 2011</th>
<th>WIFAR March 2012</th>
</tr>
</thead>
<tbody>
<tr>
<td>Wind</td>
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<td>5-7.5 m/s from NW</td>
<td>5 m/s from N-NE</td>
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<tr>
<td>Swell</td>
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<td>&lt; 1 m from NW</td>
<td>0.75-1.5 m from S-SW</td>
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<td>1-2</td>
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</tr>
<tr>
<td>Ice edge</td>
<td>Compact On-Ice wind</td>
<td>Diffuse Off-Ice wind</td>
<td>Compact On-Ice wind</td>
</tr>
</tbody>
</table>

**Table 1: Sonobuoy experiments and environmental conditions in the Fram Strait 2010-12.**

Acoustic data were recorded over approximately 3 hr periods on each of the deployment days at hydrophones of bandwidth 3 kHz at depths of 400 ft or 1000 ft. The time series were processed using a scheme similar to that used in 1987 [2]. Reported here is median noise spectral density level (NSL) spectra in dB re 1μPa²/Hz and in 1/10-octave averaged frequency bands from 10 Hz to 1000 Hz. Signal processing applied Welch spectrum estimation (1 s samples, Hamming windowed with no overlap, fast-Fourier transformed at 1 Hz frequency resolution, 10 s time-average). Estimates of median noise power levels were formed over the duration of the recordings from buoys within each of four identified ice zones (open ocean, diffuse ice, compact ice, inner compact ice) before conversion to NSL. For additional analysis, 10-s noise samples from individual buoys in different ice zones were plotted along two frequency axes. Such scatter plots (soundscape
plots) have been used for the identification of noise field components due to wind and rain, near and distant shipping, and ice conditions [9,10]. The goal here was to investigate the use of scatter plots in determining the ice composition above the hydrophone.

4. RESULTS

Figure 3 shows median noise spectra based on data from 3 to 5 buoys (depth 400 ft) in open ocean (left panel) and in inner compact ice (right panel), in October 2010 (red) and March 2012 (blue), respectively. Highest noise levels are observed for the 2010 data set at all frequencies, both in open ocean and in compact ice. The difference is significant: at 100 Hz, the NSL in open ocean is 6 dB higher in 2010 than in 2012, in inner compact ice the difference is 10 dB. At 1000 Hz, the NSL in open ocean is 4 dB higher in 2010 than in 2012, in inner compact ice the difference is 18 dB. From Table 1, in October 2010 the significant wave height outside the ice edge was 2-3 m (sea state 5), while in March 2012 the significant wave height was 0.75-1.5 m (sea state 4). Visual observations from the aircraft in 2010 showed that swell (long ocean waves) was propagating into the ice zone, while such long waves were considerably less pronounced in 2012.

![Figure 3: Median ambient noise levels recorded on sonobuoys in different regions of the Marginal Ice Zone in October, 2010 (red) and March, 2012 (blue).](image)

The higher noise levels observed in open ocean in 2010 than in 2012 can be attributed to the difference in sea state. The high noise levels in compact ice observed in 2010 can be attributed to swell propagating from the open ocean into the ice pack. During periods with swell propagating into (more or less) compact ice, individual floes deform and break up; this produces high sound levels in a wide frequency band. In 2010, this raised the noise level in compact ice to near open ocean level. In 2012, however, the noise level in compact ice is significantly lower than in open ocean. This is due to less swell propagation into the compact ice. The impact of swell propagating into the pack ice on noise levels is larger than the impact of increasing sea state (from 4 to 5) on noise levels in open ocean.

At low frequencies (below 50 Hz) significant noise contribution comes from more distant sources, both in open ocean and in compact ice. This causes the levels at low frequencies to be more similar in open ocean and compact ice, both in 2010 and in 2012. Comparing the data sets, levels at low frequencies are higher in 2010 than in 2012. Finally, the peak in noise levels at approximately 18-24 Hz, more pronounced in 2010 (October) than in 2012 (March) can be attributed to marine mammal vocalizations. Such peaks, not observed in 1985-87 data [2], were observed in recent Fram Strait noise data and attributed to fin whales (August-April) and blue whales (August-November) [3].
Fig. 4: Scatter plots of noise samples from three hours of data collected at sonobuoys in open ocean (blue), diffuse ice (green), and inner compact ice (red) zones of the Marginal Ice Zone of the Fram Strait in October of 2010. Axes are noise level (NSL) in dB re 1 μPa²/Hz at indicated 1/10-octave frequency bands.

Fig. 5: As Fig. 4, from data collected in the MIZ of the Fram Strait in March of 2012.

Figure 4 shows scatter plots of noise samples from three buoys in October 2010 processed in 1/10-octave frequency bands of 32/16 Hz, 100/16 Hz, 315/16 Hz, and 1000/16 Hz, respectively, on the horizontal/vertical axes. The three buoys (depth 1000 ft) were in open ocean (blue), diffuse ice (green), and inner compact ice (red), respectively. First observe that the clusters from inner compact ice separate from clusters from open ocean and diffuse ice at mid (100 and 315 Hz) and high (1000 Hz) frequencies. The clusters from open ocean and diffuse ice have considerable overlap at low (32 Hz) and mid frequencies, while at high frequency all clusters separate. Furthermore, we observe relatively small spread of noise levels on buoys in diffuse ice and in open ocean at 315 and 1000 Hz; the largest spread is at 32 Hz and at 100 Hz at all buoys. There is no apparent correlation between high/low levels at any frequency or ice zone.
Figure 5 shows scatter plots from three buoys in March 2012 (depth 400 ft) in open ocean (blue), diffuse ice (green), and inner compact ice (red), respectively. For this data set there is considerable overlap between the clusters from the three zones at low frequency, all clusters separate at mid frequencies, while at high frequency the open ocean and diffuse ice clusters partially overlap. The clusters from inner compact ice separate from other clusters at mid and high frequencies. Furthermore, we observe less spread at mid frequencies than at 16 and 32 Hz (all buoys), with no apparent correlation between high/low levels at any frequency. The main observations from these scatter plots are that from 2010 data, clusters from compact ice separate from other (diffuse ice and open ocean) clusters at mid and high frequencies, while clusters from open ocean and diffuse ice separate only at high frequency. From 2012 data, clusters from compact ice separate from other clusters at mid and high frequencies, while clusters from open ocean and diffuse ice separate at low and mid frequencies.

5. SUMMARY

This paper presented measurements of ambient noise in the Fram Strait recorded on fields of sonobuoys deployed in areas (size 150 km by 150 km) of the Marginal Ice Zone under different ice, weather, wind, and ocean conditions on two days in October 2010 and March 2012. Median noise level spectra (10 Hz to 1 kHz) were used to observe level dependence on sea state, wind and ocean swell properties. The main observation is that swell propagating into the pack ice raises noise levels within the ice zone more than predicted by the noise level increase due to sea state in open ocean. This is in correspondence to observations made during the MIZEX campaigns [2]. Scatter plots of noise samples from buoys in different ice zones revealed a means to discern compact-ice locations from diffuse-ice and open-ocean locations. The distinction between open-ocean and diffuse-ice conditions, however, is dependent on wind and wave forcing effects raising the noise level in diffuse ice to and above levels in the open ocean; the distinction was shown to depend on frequency. A spectral peak at 18-24 Hz observed both in the 2010 (October) and 2012 (March) data sets can be attributed to whales. Noise due to seismic exploration was not observed in these two data sets; year-long recordings on Fram Strait moorings [3] as well as on ACOBAR moorings show that such noise can be present most of the year.

ACKNOWLEDGEMENTS

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REFERENCES


Abstract: Underwater acoustic measurements have been recently carried out in Tethys Bay (Ross Sea, Antarctica) during the XXIX Italian Antarctic Expedition to investigate the environmental noise and to support acoustic propagation studies in the area. Tethys Bay is a small, deep cove close to the Antarctic Italian base Mario Zucchelli Station (Baia Terra Nova -74°42′ S e 164°07′ E), covered with sea-ice for most of the year. During the period of the experiment (November 2013) the pack-ice had an almost constant thickness of about 2.2 m, so that the measurements were performed deploying the instruments into the sea from holes having 1.3 m diameter drilled in the pack ice. They were located along the bay axis at a distance of about 500 m each other. The sea depth was around 200 m except for the hole close to the coast, were it was only 25 m. An hydrophone RESON TC 4032 was located in the outermost hole and measurements were collected at 0, 20 and 45 m depth. The measurements were repeated each time moving the acoustic source, a transceiver transmitting FSK pulses at 11 kHz, in the other three holes. During the experiment, sound speed profiles, sea temperature and salinity, currents, as well as the main meteorological parameters were continuously measured. The acquired passive acoustic measurements evidenced that the signal was generally dominated by different sounds from seals. Finally, the collected data-set and results from preliminary analysis of sound intensity attenuation is presented. The matching between the measured data and data obtained through numerical ray-tracing models of the under-ice acoustic propagation is discussed, pointing out the physical parameters that primarily impact on the attenuation.

Keywords: Acoustic measurements, Antarctica, Ambient noise, Propagation modelling
1. INTRODUCTION

Remote acoustic monitoring systems are of great interest for marine environmental research in the polar areas as they allow long-term observations even during winter, when the ice covers the sea surface or where the passage of large icebergs poses a serious risk for the instrumentation deployed in the upper layers.

In order to investigate the environmental noise and to support acoustic propagation studies, underwater acoustic measurements have been carried out in November 2013 in Tethys Bay (Ross Sea, Antarctica) during the XXIX Italian Antarctic Expedition in the framework of the Italian Program of Antarctic Research (PNRA).

Preliminary results from passive acoustic measurements and numerical acoustic propagation simulation are here described. The paper is organized as following: in section 2 a brief description of the experiment and the environmental conditions of the area is provided. In section 3 environmental noise is investigated and its main components analyzed. Finally, in section 4 measurements of transmission loss are compared with predictions by numerical model.

2. EXPERIMENTAL SETUP

Thetis Bay is a small cove located in Terra Nova Bay, close to the Italian Antarctic Base M. Zucchelli Station (74°42’ S; 164°07’ E). It is about 1600 m wide and 3000 m long with the axis oriented to the Northwest and reaches a maximum depth of 280 m in the central part, see Fig. 1 (left). It is surrounded by steep rocks and glaciers while the coast line is characterized by rocky cliffs. The seafloor is primarily granitic rock, with softer substrates composed of coarse sands or gravels. During the experiment the Bay was covered by a 2.2 m layer of sea-ice whose thickness was frequently checked because the area is used for the landing of aircrafts. Sea-ice coverage extended far from the bay to the open sea.

![Fig. 1: (left) The area and bathymetry of the experiment with the location of the measurement points. (right) Sound speed profiles in P1, P2 and P3 on 11/11 and 18/11 (dotted lines) and the resulting average profile (heavy line).](image-url)
Acoustic and oceanographic measurement were performed deploying the instrumentation through 1.3 m diameter holes drilled in the pack. The used acoustic device was composed of a RESON TC4032 hydrophone and an acquisition system sampling at 100 kHz frequency. Each record lasted 30 s. Sea temperature and depth were continuously checked by two sensors deployed close to the hydrophone. The acoustic source was a transceiver transmitting FSK pulses at 11 kHz manually operated.

The experiments were performed from three holes (P1, P2, P6) located along the axis of the Bay at a distance of about 500 m each other with sea depth varying from 25 to 250 m. The hydrophone was located in the outermost hole (P3) while the acoustic source was moved from one location to the others with the transducers deployed at 15 m. Both active and passive measurements were performed with different geometries and repeated at few days interval. CTD profiles were taken from all three sites while time series of currents, temperature and salinity at 50 m depth were obtained by leaving autonomous instruments in P3. For the analysis of environmental noise, an extended data set was used containing all acquisitions made in different locations of the bay spanning from Cape Washington to the Italian Antarctic Base M. Zucchelli Station.

Computed sound speed profiles [1] are characterized by a linear increase with the depth, from about 1439 m/s at surface to 1443 m/s at 230 m, see Fig. 2 (right) and they did not differ significantly from each other more than 0.5 m/s. Mean current was not strong enough to affect the position of the hydrophone or the source and the weather conditions were never so harsh to limit the possibility to carry out the experiments.

3. ENVIRONMENTAL NOISE ANALYSIS

Ambient noise is a composite mosaic of sound patterns that differ each other by emitting sources. In coastal areas and in very urbanized basins, noise produced by anthropogenic activities is usually the main component of underwater ocean noise, and sometimes, it masks the contribution of physical phenomena, especially wind and rain, as well as biological emissions making difficult their detection. Thetis Bay, on the contrary, is a very suitable area to listen to the ocean sound in an uncontaminated environment with very limited human interactions.

In this area, seal vocalizations are the predominant acoustic sources and their presence is well evident in quite all samples acquired during the period of the experiment. Several species of Antarctic pinnipeds live in the area, specifically Weddell seal (Leptonychotes weddellii), Crabeater seal (Lobodon carcinophaga), Leopard seal (Hydrurga leptonyx) and Ross seal (Ommatophoca rossii).

Acoustic samples were acquired in November, in the mating period for Weddell seals [2]. Most frequent calls belonged to this species and to Crabeater seals, whereas no vocalizations of Leopard or Ross seals were identified. All records were analyzed in order to determine the background noise and to classify the underwater call types of Weddell and Crabeaters seals by means of their spectral features and the identification of known sound patterns following the categorization proposed by [3]. From the overall dataset consisting of more than one hundred of acquisitions, only records without any ping of the 11 kHz acoustic source and acquired when no human activities occurred nearby the deployment area were considered for the environmental noise analysis. Samples acquired where men were walking on the pack or snowcat was in operation were discarded. Since the presence of ice pack prevent the noise due to wind forcing and the Italian Antarctic station in Thetis Bay is the only (very poor) source of anthropogenic noise, the environmental noise of the bay is due
only to biological or geophysical sources. In order to identify the effective background noise, the analysis was performed selecting only records without vocalizations, acquired in different area of the bay with the hydrophone deployed at variable depths ranging from 20 m down to 43 m.

The average background noise spectrum in the frequency band 0-10 kHz shows extremely low values well below sea state 1 in almost whole analyzed frequency range (Fig 2). The spectral analysis evidenced also the wide range of vocalizations type made by Weddell seals and allowed the detection of very common patterns such as the typical call of adult male specimen used for territory defence in mating period known as Trill (W1 and W9 in [3]). Trill was the most frequent sound pattern detected in the samples followed by series of Whistle ascending and Whistle descending (corresponding to W2, W3, W4 in [3]) and Chug (W6, in [3]). Spectra of Trills, Whistle ascending, and Whistle descending are clearly detectable compared to background noise for both greater levels and features (Fig. 2).

Trill spectra show similar characteristics to background noise up to 500 Hz whilst they began to diverge up to 1 kHz and then tended to approach again the background noise at the higher frequencies of the spectrum. Whistles descending exhibit a sharp peak just above 2 kHz, a lowest peak at about 3 kHz and a smooth decreasing towards loudest sound levels for frequencies greater than 5 kHz. Whistles ascending show a specular shape at about 4-6 kHz.

Fig. 2 Average spectra of background noise (black), Trill(cyan), Whistle descending (green) and Whistle ascending (red) compared to theoretical sea state 0, 1, 4 (grey dashed lines).

Fig. 3: Spectrogram of typical Weddell seal calls in dB re 1μPa²/Hz observed during the passive listening. From left to right: Trill, Whistle ascending and Whistle descending.
Weddell seal vocalizations are clearly identifiable by the analysis of their spectrograms (Fig. 3) since the patterns are very different in frequency band, shape, duration and rhythm. Trill calls show a descending pattern, are emitted once, last for 15 seconds and cover a wide frequency range from 6 kHz down to few hundreds of Hertz, whereas Whistles ascending, although being single pulses, last few seconds maximum and their patterns are characterized by a sharp increasing from about 4 kHz up to 5 kHz followed by a smooth rise up to 6 kHz maximum. On the contrary, Whistles descending are a series of pulses initially emitted at about one second interval progressively reducing the interval and dropping from 10 kHz to 2 kHz

4. COMPARISON OF TRANSMISSION LOSS WITH SIMULATIONS

The availability of an acoustic source (AS) has permitted to obtain a direct measure of the transmission loss in the same underwater environment in which the ambient noise analysis has been carried out. The AS transmits a narrowband signal centred at 11 kHz with source level 192 dB re 1μPa² at 1 m. The transmission is omnidirectional in the lower hemisphere. During the experiments the AS has been positioned 15 m deeper in three locations (P1, P2, P6) along the axis of the Bay, while the receiver (R) was always in the P3 station at different depth values. The actual transmission loss (TL) has been obtained by subtracting the received level (RL) in dB re 1μPa² to the source level. The observed TL values for the executed experiments are reported in Table 1. They show a constant decay with increasing distance from the source without a significant variation with the receiver depth. Only for the receiver at 23 m depth a departure from the other values occurs. Comparing the TL values with those obtained by assuming spherical spreading with Thorp’s equation for absorption loss it results a good matching for the experiments with AS positioned in P6 and P1. However, at lower range the adherence is not so tight.

The measured values of TL for the full set of AS-R geometries has been compared with those predicted by a well-established numerical method: the BELLHOP algorithm [4], a ray-tracing propagation model. An estimation of TL has been obtained by simulating the same environmental conditions found during the experiments. The set up for the simulations was the following: a ice layer of 2.2 m with flat lower surface, a monotone bathymetry profile from the AS to the R depth and the average sound speed profile in Fig. 1. Values of incoherent TL at R obtained by the simulations are in Table 1 and a full TL analysis for the AS coastal position is shown in Fig. 5 (left). The TL obtained by simulations generally underestimate the experimental values with a steady bias of about 4 dB and worst performance for receiver at 23 m depth. This difference may be explained by a limited knowledge of the boundaries properties that for specific ray trajectories can influence the TL with a major impact. The ray tracing for AS in P6 and R with depth set to 23 m is shown in Fig. 4 (left).

Finally, by simulating a free-of-ice surface in quiet sea state conditions it has been possible to evaluate the ice canopy impact on the predicted TL value. In Fig. 4 (right) it is shown a comparison between the TL curves for the experiments in Table 1 with receiver at 23 m depth. One notes that in absence of the ice canopy a decrease of TL is evident, with consequently deterioration of the model prediction. A good adherence was found between measured and predicted values with a major influence of environmental parameters with augmenting size of investigated area.
Table 1: Experiments description and simulation results.

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<td>~72</td>
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<td>~64</td>
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<td>44</td>
<td>~129</td>
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<td>~61</td>
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Fig. 4: Simulations for receiver at 23 m depth. (left) Transmission loss and ray trajectories with source in P6: direct path (red), with (black) and without (green) ice reflections. (right) Transmission loss curves: ice canopy (heavy line), spherical spreading with Thorp’s attenuation (dashed line), no-ice canopy (dotted line) and measured value (×).

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REFERENCE

SOURCES OF LONG-TERM AMBIENT OCEAN SOUND NEAR THE ANTARCTIC PENINSULA


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Abstract: Hydrophone arrays (250–1000 Hz) were deployed within the Bransfield Strait and Scotia Sea (Antarctic Peninsula region) from 2005 to 2009 to study sources of ambient ocean sound. Icequakes, which are broadband, short duration signals derived from fracturing of large free-floating icebergs, are a prominent feature of the ocean soundscape. Icequake activity peaks during austral summer and is minimum during winter, likely following freeze-thaw cycles. Iceberg grounding and rapid disintegration also releases significant acoustic energy, equivalent to large-scale geophysical events. Overall ambient sound levels can be as much as ~10–20 dB higher in the open, deep ocean of the Scotia Sea compared to the relatively shallow Bransfield Strait. Noise levels become lowest during the low annual temperatures of the austral winter, likely due to freezing of regional sea ice of all scales. Ambient noise levels are highest during austral spring and summer, presumably due to melting and cracking icebergs. Vocalizations of blue (Balaenoptera musculus) and fin (B. physalus) whales also dominate the long-term spectra records in the 15–28 and 85 Hz bands. Blue whale call energy is a maximum during austral summer-fall in the Drake Passage and Bransfield Strait when ambient noise levels are a maximum and sea-ice cover is a minimum. Fin whale vocalizations were also most common during austral summer-early fall months in both the Bransfield Strait and Scotia Sea. The hydrophone data overall do not appear to show sustained anthropogenic sources (e.g. ships), likely due to low coastal populations and the difficult marine conditions of the Southern Ocean.
Keywords: ocean ambient sound, ice noise, seasonality, baleen whales

INTRODUCTION

The climate of the Antarctic Peninsula is the most rapidly changing in the Southern Hemisphere, with a few degrees Celsius rise in both atmospheric and surface ocean temperatures over the last few decades\(^1\)-\(^3\). Associated with this ongoing warming is a cycle of ice sheet and iceberg breakup and grounding that is accompanied by the release of acoustic energy into the Southern Ocean. Although much attention has been given to the increasing anthropogenic contributions to ocean noise, which may be as much as 12 dB over the last several decades\(^4\), the sounds created by ice breakup at the poles may represent an underappreciated, yet significant, natural contribution to the ocean noise budget.

To study these ice-related and other sources of natural sound, we deployed arrays of hydrophone moorings in the Bransfield Strait, a protected sea along the western Antarctic Peninsula, and the Scotia Sea, an area of the South Atlantic northeast of the Antarctic Peninsula. A variety of cryogenic sounds were recorded, from iceberg grounding and breakup to sea-ice “quakes.” Icequakes are impulsive broadband signals with durations between 10 and 80 sec and dominant energy over the ~10–125 Hz band. Other sources of low-frequency sound are vocalizations of large baleen whales, which dominate the 15–28 and 85 Hz frequency bands\(^5\),\(^6\) and the waterborne acoustic phases of seafloor earthquakes\(^7\). The goal of this paper is to present the various sources of ambient sound in these regions and estimate long-term variability in acoustic energy levels to identify what processes characterize and influence the soundscape in this part of the Southern Ocean.

HYDROPHONE INSTRUMENTATION

In order to monitor the levels and determine the sources of acoustic energy near the Antarctic Peninsula, two arrays of hydrophone instrument moorings were deployed in the Bransfield Strait and Scotia Sea (Fig. 1). The Bransfield Strait arrays consisted of five and six hydrophone moorings deployed December 2005–2007 and November 2008–December 2009, respectively\(^7\),\(^8\). One hydrophone mooring was deployed in the Drake Passage, north of the Bransfield Strait, during December 2005–2006. A total of five hydrophones were deployed in the Scotia Sea from 2007 to December 2008.

The autonomous hydrophone instrument package consists of a single ceramic hydrophone, a filter/amplifier stage, an accurate clock, a low-power processor, and a battery package. The instrument is capable of recording at 16-bit data resolution between 250 and 1000 Hz (phone in Drake Passage) for periods of 1 to 2.5 years, respectively. The pre-amplifier is designed to equalize the spectra against typical ocean noise over the pass-band with an 8-pole anti-aliasing filter. A micro-processor controlled, temperature-correcting crystal oscillator with an average time drift of 400 ms yr\(^{-1}\) provided accurate timing during the typical 1–2 year deployment duration. The electronics are housed in a titanium pressure case rated for 1200 m depths. The instrument case is attached to an anti-strumming oceanographic mooring with anchor, acoustic release, mooring line, and a syntactic foam float to place the sensor at depths of 300–500 m.
Fig. 1: Maps showing icequakes (black dots) and geographic location of the Antarctic Peninsula (AP), Bransfield Strait (BS), Drake Passage (DP), and Scotia Sea. Yellow triangles show 2005–2006 hydrophone mooring deployments within the Bransfield Strait (elongated basin northwest of AP) and Drake Passage, as well as the 2006–2007 deployments within Scotia Sea. Circles M2 and M4 show locations of hydrophone moorings used to make spectrograms in Fig. 4. Red line is satellite track of iceberg A53a through arrays during early 2008. Outline of iceberg shown for scale. Red dot is location of wind station in Fig. 5.

SOURCES OF AMBIENT SOUND

Figure 2 shows the spectrograms of the most common acoustics signal sources recorded on the Bransfield Strait and Scotia Sea hydrophones, which vary from biological to geophysical to cryogenic. The signals (Fig. 2a,b) that dominate the record are the broadband acoustic arrivals of icequakes from the breakup of large-scale sea ice. Icequakes can be emergent (Fig. 2a) or impulsive and short duration (Fig. 2b), and it seems both signal types are likely due to source affects at the iceberg\[9\]. Figure 2c shows the fundamental frequency and harmonic overtones of tremor from a grounding iceberg. Figure 2d shows the distinct Antarctic blue whale call, characterized by a repeating 28 Hz tone plus downsweep\[6\]. Figure 2e shows the acoustic coda of a seafloor earthquake (referred to as a “T-wave”) with broadband signals of an airgun in the background. Lastly, Fig. 2f shows the characteristic sound created by a ship, generated by cavitation from the propeller. The sound energy level is roughly equivalent for all sources, but each varies in prevalence through the year.
Fig. 2: Spectrograms of the varied acoustics sources recorded on hydrophone arrays. (a) emergent icequake; (b) impulsive, short duration icequake, indicating that icequake may be closer to receiver; (c) fundamental and harmonic overtones of iceberg tremor; (d) Antarctic blue whale vocalization; (e) is an earthquake (T-wave) packet with broadband airgun arrivals in background; and (f) is noise from a ship, generated by cavitation from the propeller. The received sound energy levels appear roughly equivalent for all sources; however, each varies in prevalence through the year.

LONG-TERM AMBIENT SOUND SOURCES AND LEVELS

Long-term spectrograms of the acoustic power spectral density and the daily average sound levels recorded on hydrophone M2 in the Bransfield Strait and M4 in the Scotia Sea are shown in Figs. 3 and 4. Since the Bransfield array was a three-year deployment as compared to a one-year deployment in the Scotia, the signals in Fig. 3 appear much more compressed in the Bransfield spectra. For consistency in comparison, the frequency-dependent hydrophone instrument response was removed from both spectra prior to display.

From Fig. 3, overall noise levels appear to be higher in the Scotia Sea than the Bransfield Strait. The near constant, broad-band, short-duration impulsive signals present in both spectra in Fig. 3 during the austral summer to fall (December to June) are icequakes, or ice-breakup sounds, likely due to increased summer temperatures, and wind-driven ocean waves. There is also continuous, low-frequency energy present in both regions. This energy is focused under 5 Hz in the Bransfield, and under 10 Hz in the Scotia. During several periods of time, which can last from weeks to months, the energy develops into tremor-like signals with a fundamental at 1–3 Hz and several overtones. We interpret this low energy to be from a combination of sources, including broadband energy created by sea-state (storms, waves, and wind), minutes-to-hours long periods of iceberg grounding tremor, as well as periods of tonal “strumming” caused by fast moving ocean currents that make mooring line vibrate. Although the icequakes show a clear seasonal variation, this low-frequency energy remains present
throughout the year. This is consistent with the idea that this energy is caused by currents, storms, and grounding icebergs that are present year-round in the Southern Ocean.

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**Fig.3**: Long-term spectrograms of the acoustic power spectral density in units of dB re 1 μPa²Hz⁻¹ at the Bransfield Strait (top) and Scotia Sea (bottom). Spectrograms are calculated from 2 second FFT windows with each vertical time slice representing the cumulative energy over a 1 day interval.

The acoustic energy focused in the 15–28 and 85 Hz bands (Fig. 3) are fin whale calls\[^{5,10}\]. The steady signals within the 28 Hz band are the vocalizations of Antarctic blue whales\[^{5,6,10}\], which overlap in time with fin whale calls. The blue whale vocalizations (most easily seen in the Scotia Sea data) are characterized by a long duration (~10 sec) 28 Hz tone, infrequently followed by a downsweep to 19–20 Hz\[^{6}\] (Fig. 2d). However, the long-term spectrogram (Fig. 3) shows the blue whale signal energy is predominantly focused in the 28 Hz band with little downsweep energy apparent. Blue whales can also generate overtones up to 80 Hz that are occasionally seen in the spectra but are apparent during February 2006 in the Bransfield data. The fin whale vocalizations show a clear seasonality and reach a maximum during the austral fall-winter months, following the peak in icequake activity. The blue whale calls are, however, present nearly year-round (this can be most readily seen in the Scotia record), which is also consistent with the previous findings\[^{5}\]. As can be seen from Fig. 3, the vocalizations of these whales are a significant, nearly continuous, component to the ambient sound field in this region.
Fig. 4: Diagram shows selected percentiles from the cumulative distribution of spectral energy (daily average) recorded on hydrophone M2 in the Bransfield Strait (2005–2007) and M4 in the Scotia Sea (2008–2009). Solid line shows the dB levels of 50% of the cumulative spectral energy in the given frequency band; lower dashed line shows the dB level of 5%; and upper dashed line shows the dB level of 95% of the cumulative spectral energy.

The difference in sound levels between the Scotia Sea and the Bransfield Strait can be more clearly seen in the comparison of sound levels between the two regions in Fig. 4, which shows selected percentiles of the long-term, cumulative distribution of spectral energy (daily average) recorded on hydrophones M2 (Bransfield Strait) and M4 (Scotia Sea). The noise levels were separated into four distinct frequency bands to examine the influence of different sound sources on overall levels, with the 3–10 Hz band for iceberg tremor, and to avoid potential mooring line strumming noise (from fast ocean currents), which is <3 Hz, 11–30 Hz for blue and fin whales, 31–50 Hz for icequakes, and 51–90 Hz for a combination of icequakes and fin whale pulses. The three lines shown in Fig. 4 represent the sound level (dB) distribution for that day in each frequency band, where the solid line shows the 50%, or median, distribution, where 50% of the sound energy falls below this level during the given day. The lower dashed line is the 5% level and the upper dashed line shows the 95% level. The Scotia Sea ambient sound levels were not overlain with the Bransfield Strait data because the data sets are not time synchronous. Nonetheless, the Scotia Sea noise levels show similar seasonal variations as the Bransfield Strait and Drake Passage levels. The 51–90 Hz bands have peak energy during austral fall-winter (May–June), likely due to an increase in sea-ice cover and contributions from peak blue and fin whale vocalizations during this time. The downward trend in noise levels leading to a minimum during August (late austral winter) also mirrors the minimum levels observed in the Bransfield Strait and Drake Passage, likely due to freezing, or “locking,” of sea ice, in that it does not break apart and create sound. In contrast, all frequency bands show a significant rise in noise levels during spring-summer (September–December), possibly a result of increased ice melt due to seasonal increases in
ocean and air temperatures. Even though the Bransfield Strait and Scotia Sea show similar variations in annual noise levels, the sound levels are from 10 to 20 dB lower in the Bransfield than the Scotia in all frequency bands.

Figure 5 shows ambient noise levels averaged over 51–90 Hz from one hydrophone in the Bransfield Strait and one in the Drake Passage. We chose this band to minimize the blue and fin whale call energy contribution, in an attempt to highlight the ambient noise levels due to ice, meteorological, and geophysical (non-biological) sources. One caveat is that this band still includes the 85 Hz fin whale overtone, which will influence ambient sound levels during peak calling months. The sound average from the Bransfield Strait and Drake Passage was then compared to the daily record of air temperature and wind speeds measured at King George Island, located between the Bransfield Strait and Drake Passage, illustrating the relationship between environmental factors and noise levels. Interestingly, ambient noise levels become lowest in the Bransfield Strait during the lowest annual temperatures of the austral winter months. Also, there is a correlation between wind speed and noise levels, although this correlation is weaker than with temperature. We interpret the low noise during this period as due to the freezing of sea ice, where little to no ice breakup is occurring. Noise levels are highest in the Bransfield Strait during austral fall, presumably when the amount of sea ice and sea-ice cover is increasing, causing more noise. The Drake Passage does not show as large a variation in noise levels as the Bransfield Strait, possibly due to its lack of extensive ice cover\(^ {11}\). However, the highest noise levels in both the Bransfield Strait and Drake Passage correlate with the times of the highest recorded wind speeds during the austral fall-winter (May–September). Thus, it seems the wind induces high noise conditions by increasing wave heights\(^ {4}\), thereby facilitating sea-ice breakup during the fall; but as air temperatures decrease through the winter, large patches of sea ice become frozen, effecting a reduction in noise caused by ice breakup and wave action.
Although both ships and airgun (for research and/or oil exploration) sounds are occasionally present on the hydrophone data (Fig. 2e,f) we do not see a sustained contribution to the long-term noise spectra from these anthropogenic sources as is seen in the north Atlantic and European Arctic\cite{12}. It is well established that there are much greater levels of ship traffic in the Northern Hemisphere as compared to the Southern Hemisphere\cite{4}. The lack of ships and oil exploration in the Southern Ocean is easy to understand given the low coastal populations, generally hazardous marine weather, and presence of sea ice. The lower density of ship traffic has been used to explain the ~20 dB lower average noise levels observed at some Southern Hemisphere sites\cite{13}.

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REFERENCES


Session 6

Acoustics of Bubbles, Oil and Gas

Organizers: Tim Leighton and Lee Culver
A TECHNIQUE TO MEASURE THE REAL SURFACE TENSION ON A BUBBLE WALL

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Abstract: The surface tension of water is a key parameter for assessing the degree of contamination of harbours, ports and open waters. However standard methods of measuring surface tension do so at the flat air/water interface at the top of a body of water, whereas in for many important processes, this is not the manifestation of surface tension that is most important. For biogenic decomposition, air/sea gas exchange, the production of aerosols (all three of which are known to be important for climate on a global scale), as well as the behaviour of ship wakes and their effects (in distributing contaminants in the water column, affecting the local sound field or the ship’s acoustic signature), and in the harvesting and transportation of petrochemicals, it is the surface tension on the wall of bubbles within the water column that matters. This paper explores a technique that can measure the surface tension on a bubble wall, and compares it with the surface tension measured at the air/water interface. Any difference would mean that modelling of the above effects, based on measurements of surface tension on the flat air/water interface, would contain systematic errors with global implications.

Keywords: bubble, surface tension, air/sea gas flux, pollution, wake
1. INTRODUCTION

The question ‘what is the surface tension on a bubble wall’ is key to many environmental processes, including: the dynamics of bubble clouds under breaking waves and in ships wakes and their effect on sonar and air/sea gas transfer; and the formation of aerosols at the sea surface and their subsequent effect on climate; and the transport of pollutants.

However the actual meaning of the question ‘what is the surface tension on a bubble wall’ is less straightforward than might first be thought. There is a parameter ($\sigma$) that we call ‘surface tension’, that enters formulations the idealized physics of a range of processes. These include the balance of forces when a liquid film breaks, the formation of liquid droplets when such films break, the coalescence and fragmentation of liquid drops and or gas bubbles in liquids, the motion of capillary waves on a gas/liquid interface, and many more. They enter these formulations because the ‘surface tension’ can be incorporated into the idealized physical model of the process through its twin abilities to equal the energy required to form a unit surface area of new interface, and to equal the force per unit length perpendicular to an imaginary line drawn in a flat interface. However these simple equalities leave out a wealth of detail that escapes the current physical models of these process, the most obvious being whether the process is done in static conditions (i.e. so slowly as to an equilibrium state to be established in the surface), or in dynamic conditions (i.e. so rapidly as to prevent equilibrium conditions from being established).

This difference means that a measurement of the ‘value of the surface tension’ undertaken by one accepted technique, may not be transferrable to another process. The implication is that textbook values of surface tension, or values of the surface tension of a particular water sample measured using accepted techniques in the field or the laboratory, may not be accurate to extrapolate to predict how that same water sample will behave when it participates in environmentally- and climatically-important processes.

This paper will report on the preliminary measurements made to explore these issues by comparison of single ‘look-see’ measurements (repetitions and statistical analysis will be made when enough data have been compiled) of the value of surface tension on a bubble wall using the threshold for the formation of Faraday waves, and measurement of the value of surface tension in the same liquid sample using a standard method.

2. BACKGROUND

The well-known du Noüy ring tension meter method is used in this paper to obtain values of surface tension that an accepted example laboratory method would produce. A ring is immersed in the liquid [Fig. 1(a)], and then the vertical separation between the ring and the liquid is increased, and the force on the ring measured, until the ring separates from the liquid [Fig. 1(b)]. The vertical separation is increased sufficiently slowly that equilibrium conditions exist on the surface.

To compare with this, we obtain an estimate of the surface tension that pertains to the threshold for the excitation of Faraday waves on the bubble wall [1, 2]. Once the acceleration of a pulsating bubble exceeds a threshold value (which usually means that the amplitude or frequency of the sound field driving the bubble to pulsate has passed some
critical value [3]), after a short transient period [4, 5] that then leads to a steady-state pattern being formed on the bubbles wall [6, 7], a perturbation is superimposed on the bubble wall motion which corresponds to that spherical harmonic which (i) had non-zero order and (ii) has a natural frequency that is closest to half of the driving frequency. This is called the Faraday wave, first characterised by Faraday for flat air/liquid interfaces [8-10] and since studied on liquid drops [11]. If, say, the amplitude of the driving sound field continues to increase, additional spherical harmonic perturbations will be excited, and superimposed on the bubble wall motion [12].

Fig. 1: (a): Photograph of the well-known du Noüy ring tension meter method used for comparisons in this paper. (b): Frame showing the moment as the liquid film breaks away from the ring of the du Noüy ring tension meter (the white line at the bottom left of the frame is a 1 cm long scale bar).

Fig. 2: A series of air bubbles rise through a layer of salt water, 5% (w/v), coloured by food dye, to then rise through the interface (labelled ‘i’) into an upper layer of pure water. The wake (labelled ‘w’) is visible because the upper bubble has carried with it material from the coloured salt water layer.
The generation of a Faraday wave can be detected by electrochemical techniques [13-15], and by acoustical [16-24] and optical imaging [25], and it was proposed that these measurement points could be inverted to infer the value of surface tension, as it pertains to the threshold for Faraday waves. The environment around a bubble wall is known to be dynamic. Bubble growth, through coalescence, or fragmentation/dissolution mechanisms, which reduce the bubble size, are likely. In addition the exact conditions at an individual bubble wall may differ significantly to those at the gas/liquid interface of the bulk fluid which is determined by the du Noüy ring tension meter. Hence the ability to determine the surface tension on a bubble within the bulk of the fluid is highly advantageous particularly because a bubble can collect surface active agents (and indeed particulate matter) on its wall as it moves through a liquid (Fig. 2).

![Fig. 3: The apparatus used for measuring the Faraday wave threshold.](image)

3. EXPERIMENTAL PROCEDURES

The apparatus shown in Fig. 3 uses a 40 kHz sound field. It is assumed for the moment that this levitating field is sufficiently far from the bubble resonance that it does not contribute to the acceleration of the bubble wall sufficiently to affect the threshold for Faraday waves. Unless resources are available to conduct these experiments in a microgravity environment [26-28], the bubble must be held at the focus (and in the field of view) of the camera, and the levitation method [29-35] is less invasive than tethering under glass bodies, or with hoops of fine wire to thread [36, 37].

The use of mirrors ensures that the bubble is properly illuminated and that images of it from top and side are simultaneously in focus in each frame of a high speed camera (as is the ruler). The experiments must be done quickly to avoid rectified diffusion, and so a
preprogrammed increments of increasing drive field amplitude are run until the faraday wave is detected, and then the frequency is increased and the process repeated.

A monolayer of decane \((\text{CH}_3(\text{CH}_2)_8\text{CH}_3)\) was placed on the surface of the previously fresh, purified water, and the value of the surface tension was measured in three ways: (i) using the du Noüy ring tensiometer meter; (ii) using the Faraday wave technique on a bubble injected into the body of the water; and (iii) using the Faraday wave technique on a bubble entrained into the body of the water through the upper liquid/air interface (and the decane monolayer) by a water droplet impacting the water body from above.

Fig. 4: The results of measuring the value of the surface tension using the du Noüy ring tension meter (blue column); and using the Faraday wave technique on a bubble injected into the body of the water (red column) and a bubble entrained through the decane layer by a water droplet. These represent the preliminary findings from a single experiment and have yet to be repeated. The error bars reflect the estimated limits of precision of the measurement technique. As such these can only be treated as preliminary ‘look-see’ data.

4. RESULTS

According to the preliminary data of Fig. 4, the ring method sees barely any reduction in the surface tension (from the ideal value for a pure air water interface of 72.8 mN/m at 20°C) resulting from the presence of the decane monolayer. The injected bubble appears to be best described by a marginally lower surface tension, though more data are required. The bubble entrained by a drop through the decane layer has a significantly lower surface tension.
5. DISCUSSION AND CONCLUSIONS

The du Noüy ring technique appears not to be affected by the presence of a monolayer of decane on the water interface (see Fig. 4). The surface tension measured is slightly higher than that measured using Faraday waves on the injected bubble, which could be said to be ‘clean’ as the injection apparatus was immersed prior to the addition of the decane. However, the entrained bubble using the drop method is significantly altered compared to the bubble injected into the bulk liquid. This implies that a proportion of the entrained bubble has been loaded with decane which reduces the surface tension that best describes Faraday wave motion in that case.

6. CONCLUSIONS

If a value of surface tension is to be used in a model to predict, say, the formation of bubble clouds or the injection of aerosol droplets into the atmosphere, and used as input to predict the contribution these processes have to weather or climate, it is important that the value of the surface tension that is used is critically assessed to ensure that it is derived from a measurement process in which the same properties of surface tension are characterised, as are relevant to the physical mechanisms in play at sea. This extends to measurement of ‘surface tension’ for biomedical contrast agents, food and pharmaceutical manufacturing etc.

REFERENCES


PASSIVE ACOUSTIC QUANTICATION OF GAS RELEASES

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Abstract: The assessment of undersea gas leakages from anthropogenic and natural sources is becoming increasingly important. This includes the detection of gas leaks and the quantification of gas flux. This has applications within oceanography (e.g. natural methane seeps) and the oil and gas industry (e.g. leaks from undersea gas pipelines, carbon capture and storage facilities). Gas escaping underwater can result in the formation of gas bubbles, and this leads to specific acoustic pressure fluctuations (sound) which can be analysed using passive acoustic systems. Such a technique offers the advantage of lower electrical power requirements for long term monitoring. It is common practice for researchers to identify single bubble injection events from time histories or time frequency representations of hydrophone data, and infer bubble sizes from the centre frequency of the emission. Such a technique is well suited for gas releases that represent low flow rates, and involving solitary bubble release. However, for larger events, with the overlapping of bubble acoustic emissions, the inability to discriminate each individual bubble injection event makes this approach inappropriate. In this study, an inverse method to quantify such release is used. The model is first outlined and following this its accuracy at different flow rate regimes is tested against experimental data collected from tests which took place in a large water tank. The direct measurements are compared to estimates inferred from acoustics.

Keywords: methane seeps, inverse problem, passive acoustic
1. INTRODUCTION

Assessing levels of underwater gas leaks from anthropogenic sources (pipelines, Carbon Capture and Storage facilities, etc.) or natural sources (hydrocarbon gas release) is of importance because of their economic and polluting impacts. These releases can take the form of bubbles that present specific acoustic emissions [1,2]. For the purpose of quantification of gas release, passive acoustics are applicable and it presents advantages in term of cost and electrical power consumption. Leighton and Walton were the first to use such emissions to quantify gas bubble formation in the natural world [3], and now it is common practice to determine gas volumes by relating the centre frequency of the acoustic traces to the bubble sizes [4–6]. However, this approach requires distinguishing each bubble emissions and although signal processing techniques can help as the flow rate increases [7], in essence it is limited to releases representing low flow rates [8]. The inversion scheme proposed by Leighton and White [9] is aimed at quantifying higher volume gas discharges, especially in the case where the acoustic emission of each single bubble overlaps with other bubbles. This paper draws on experimental data to assess the accuracy and applicability of the model.

An experiment, which was carried out in a large water tank facility, is presented here. Metered amounts of gas were released from an arrangement of needles and the acoustic emissions were recorded. Gas was injected in two ways: 1) with a stationary flow rate at different regimes, and 2) with varying flow rates. In each case, based on the inverse model [9], gas flow rates are estimated and compared to direct measurements.

2. EXPERIMENTAL PROCEDURES

In a large water tank (8 m x 8 m x 5m deep) of volume \( V = 320 \text{ m}^3 \), bubbles were released from an arrangement of needles (1.2 mm nozzle diameter) at the bottom of the enclosure. The bubble generation system was placed at large enough distances from the side walls so the reverberation has no effect on the damping and resonance frequency of the bubbles [10]. The gas used was nitrogen and the amount injected was controlled and metered by a mass flow meter (Bronkhorst high-tech in-flow F-111BI). The gas was injected with flow rates kept steady at 15 discrete regimes and also with varying flow rates over a period of 200 seconds.

The acoustic emissions of the bubbles were recorded using a SM2M+ hydrophone unit (wildlife acoustics). This acoustic recorder consists of a buoyant body containing an embedded data acquisition board connected to a calibrated hydrophone and is powered by internal batteries. A schematic of the experiment is presented in Fig. 1. First, similarly to Bergès et al. [11], flow rates were kept steady at 15 regimes for 30 seconds. From the spectra of the acoustic emissions at different regimes, gas fluxes are estimated. In addition, in this gas injection case, at all regimes, signals were acquired at different distances from the bubble release site in order to investigate the effect reverberation has on the results. Second, a scenario is considered where the amount of gas is varied over a period of 200 seconds. In this case, flow rate rates are estimated every second. Signals are acquired at a sample frequency of 48 kHz.
Using the method of Leighton and White [9], from the acoustic traces of the release of bubbles, the power spectral density \( S(\omega) \) is calculated. At frequency \( \omega \) this relates to the distribution of bubble sizes in the form of bubble generation rates \( D(R_0) \) as:

\[
S(\omega) = \int_0^{+\infty} D(R_0) |X_b(\omega, R_0)|^2 dR_0,
\]

with \( |X_b(\omega, R_0)|^2 \) the predicted magnitude of the Fourier transform for a bubble of radius \( R_0 \) at a frequency \( \omega \) [9]:

\[
|X_b(\omega, R_0)|^2 = \left( \frac{\omega_0^2 R_0^3 \rho_0 R_{ei}}{r - \frac{R_0}{R_0}} \right)^2 \times \left[ \frac{4([\omega_0 \delta_{tot}]^2 + 4 \omega^2)}{[(\omega_0 \delta_{tot})^2 + 4(\omega_0 - \omega)^2][\omega_0 \delta_{tot}]^2 + 4(\omega_0 + \omega)^2} \right].
\]

The quantity \( \rho_0 \) is the density of the water, \( r \) the range from the bubble to the sensor, \( \omega_0 \) the bubble natural frequency [1] and \( \delta_{tot} \) the total bubble damping coefficient [12,13]. One important unknown to consider is the initial amplitude of the bubble wall at the start of the emission \( R_{e0/l} \). This quantity can differ for example with different type of nozzles of bubble sizes. This is expressed relative to the equilibrium bubble radius and in this paper, \( R_{e0/l}/R_0 = 3.7 \times 10^{-4} \) is used following the procedure of reference [9]. This factor
constitutes the biggest source of uncertainty and its relevance has for example been discussed by Deane and Stokes [14] or Leighton and White [9]. In this study, focus is given to the effect due to the relative change in flow rates. From Eq. (1), the problem can be approximated at discrete frequencies and bubble radii and expressed in matrix form [15]. The function $D(R_0)$ can then be determined in different bubble size bins. Furthermore, as the problem tends to be ill-posed because of measurement errors, regularization in the form of a Tikhonov regularization is introduced. In matrix form, the regularised solution is as:

$$\mathbf{D} = (\mathbf{X}_b^TX_b + \alpha^2\mathbf{I})\mathbf{X}_b^T\mathbf{S},$$

with the regularisation factor $\alpha$ chosen by the mean of a Generalised Cross Validation function including a positivity constraint on the distribution of bubble sizes $D(R_0) > 0$ [15]. The inversion gives a bubble count in terms of the radius the bubble would have if it was spherical ($R_0$), since this is the parameter that governs the frequency of the sound emission [1,2]. Thus, it can be easily converted into flow rates assuming individual bubble volumes of $4\pi R_0^3/3$. This gives direct comparison to the measurements from the mass flow meter.

The starting point of the inversion scheme is the acoustic trace of bubbles in free field condition. Because here the acoustic measurements were acquired in a water tank, the reverberation of the chamber results in increased acoustic energy. It is important to investigate the effect this has on the inversion scheme. In an enclosure, there is a combination of the direct and reverberant field and the averaged total rms pressure is given by [16,17]:

$$\overline{p_t^2}(r) = S^2\left[\frac{1}{r^2} + \frac{1}{r_0^2}\right],$$

with $S = p_d \times r$ the source output, the product of the range and the direct field pressure $p_d$. The quantity $r_0$ is the radius of reverberation, corresponding to the distance where the reverberant and direct fields have equal contribution. This is given by [16,17]:

$$r_0 = \sqrt{\frac{AQ_\theta}{16\pi}},$$

with $A = 55.3 \times V/T_{60}c$ the Sabine coefficient being dependent on the sound speed in water $c$. The reverberation time in the enclosure is $T_{60} = 181$ ms between 0.8 kHz and 8 kHz (frequency band of interest in this study) and the directivity factor $Q_\theta$ is taken equal to 2, corresponding to an omnidirectional source lying on a reflective flat surface [16]. The radius of reverberation is $r_0 = 1.62$ m in the conditions considered here.
For the case of steady flow rates, measurements at 10 distances were repeated and the Sound Pressure Levels (SPL) were determined. The 10 data points at each regime are plotted against the distance from the source and a fitting based on Eq. (4) is performed. This is shown in Fig. 2 for regimes 10 and 15. The direct field decay can be observed at short ranges while the contribution of the reverberant field is noticeable at largest r (level stabilizing). Because it is needed to apply the inversion to data in free field, it is important to account for the effect of the reverberation field. To that purpose, the measurements at the shortest range (1 m) are used because at this distance, the direct field dominates over the reverberant field.

![Fig. 2: Acoustic emissions from bubbles at different regimes recorded at 10 different distances. Black circle (regime 15) and grey cross (regime 10) markers are the measurements. Black (regime 15) and grey (regime 10) lines are the best fits using Eq. (4)'](image)

3. RESULTS

First, results for steady flow rates are presented in Fig. 3 (left axis). At the 15 regimes, the spectra from the bubble emissions are inverted to obtain bubble emission size distributions at discrete size bins. From these distributions, assuming spherical bubbles, flow rates can be inferred and compared to measurements from the mass flow meter. At the highest regimes, the results from the inversion scheme correlate with the metered flow rates. Between regimes 15 and 10, the flow rates are underestimated by 1.5 dB to 3.7 dB. Because the model described in Sec. 2 relies on the quantity $R_{\epsilon_0}/R_0$ the agreement in absolute level could be improved by using a value that account for this specific bubble injection system. However, independently from the absolute level of the flow rate estimates, it can be noticed that there is an agreement with the relative decrease. Between regimes 15 and 10, flow rates decrease by 35% and this is estimated to 57% from the acoustic emissions.

The inversion method is sensitive to noise and the signal to noise ratio (SNR) is shown in Fig. 3 (right axis, dashed line). Noise floor was measured when no gas was being
injected and the ratio of measurements at each flow rates gives the SNR. The agreement of the acoustically inferred flow rates is seen to decrease with decreasing SNR. As SNR diminishes, the estimates fail to track the change in flow rate, this corresponds to regimes where the SNR is too low for the inversion to perform accurately.

Fig. 3: metered and acoustically inferred flow rates at different regimes. Left axis: mass flow meter measurements (straight line) and acoustic estimates (diamond markers). Right axis: SNR (dashed line)

Second, results from varying flow rates are presented in Fig. 4. This compares metered and acoustically inferred flow rates when the injected gas was being varied. Estimates are computed by inverting the power spectral density of the acoustic traces for each 1 second window. It can be observed that the estimates track the changes in flow rate. Also, local fluctuations of the flow rates inferred from acoustics is noticeable and is due to the influence of the background noise that corrupts the bubble count when the signal spectra are inverted. This can be mitigated by applying a filter to smooth the results.

Fig. 4: Results from data collected during laboratory experiment (Fig. 1(a)). Comparison between metered (solid black line) and acoustically inferred (solid grey line) flow rates over 200 seconds of a varying release of nitrogen gas.
4. CONCLUSION

In this study, the applicability of the inversion scheme described by Leighton and White [9] to quantify “high volume” gas leaks is demonstrated by the mean of results from a gas injection experiment in a large water tank. First, gas was injected and controlled using a mass flow meter at 15 flow rates regimes. Gas releases are estimated from the acoustic emissions and compared to metered results. A good agreement is found at high flow rates. The absolute level of the estimates is dependent on the quantity $R_{e01}/R_0$ that needs further research. The influence this factor has on the uncertainty of the estimated flow rates will be presented in separate work. Second, gas was injected and varied over a period of 200 seconds. Applying the inversion method every second allows tracking the changes in flow rates.

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ACOUSTICAL CLASSIFICATION OF THE SHALLOW SEDIMENT GASEOUS STRUCTURES IN THE SOUTHERN BALTIC SEA

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Abstract: The common occurrence of surface gas saturated sediments and gas outflows from the sea floor have been recognized relatively recently in characteristic regions of whole world. So far in the area of the Polish Exclusive Economic Zone of the Southern Baltic Sea only a few trials were taken to determine acoustically the distribution of gas saturated sediments.

In the paper, a short introduction to present-day acoustic inventory of shallow gas occurrence and its expression in the Southern Baltic Sea is presented. Signals of various frequencies and beamforming were employed during surveys conducted in 2009-2014. Simultaneous usage of several single beam echosounders applying CW signals with diverse frequency bands from 12 up to 200 kHz allowed to distinguish different forms of gas. Simple classification methods basing on variety of echo envelope parameters were applied to distinguish different forms of shallow sediment gaseous structures. Several examples of echo images and effect of their classification, obtained for different frequency bands and associated with different forms of gas existence in sediments is presented.

Keywords: methane, bubbles, parameters, classification, Baltic
1. INTRODUCTION

Manifestations of the wide-ranging and considerable quantity of the shallow gas has been identified several times in the sediments of the southern part of the Baltic Sea. However, the exact location of shallow gas-saturated sediments in whole area is known to a limited extent. Undertaken research in the Southern Baltic Sea so far have focused mostly on the deeper parts of the sea bottom, under 1 km below the sediment surface, and only a few studies have focused on the gas storage in the bottom top layer [1,2,3].

So far, observations of the gas bubbles in the top bottom layer in the Polish Exclusive Economic Zone of the Baltic Sea (PEEZ) were conducted mainly by Polish Geological Institute in the western and central parts of the area. These studies only to a limited extent allowed to draw conclusions on the occurrence of gas in the upper layer of sediment [4]. Several attempts were taken in order to observe acoustically manifestations of the presence of gas bubbles in eastern part of the region - in sediments of the Gulf of Gdansk [5,6], and in Bay of Puck [7], the areas of occurrence rich in organic carbon groundwater discharges [8], and high sedimentation rate [9].

From 2008-2014 we have made a number of attempts in order to acoustically identify forms of gas appearance in the Southern Baltic surface sediments and its distribution [10]. For this purpose a number of diverse (in terms of frequency) acoustic echo-sounders have been used simultaneously. For the fast recognition of the various form of the gaseous sediments, we have applied clustering methods based on a factors obtained during parameterization of the acoustic echo signals.

2. STUDY AREA

The main study area is situated in the Southern Baltic Sea: Gulf of Gdańsk and Gdańsk Deep, regions under strong anthropogenic influence. We observe there various geomorphological structures and a broad spectrum of sediments from the coarse-grained sands to the clayey silt. Hydrological conditions in this area are significantly influenced by the Vistula river discharge [11]. In the bottom waters it is possible to observe anaerobic conditions. In this region, occur excessively high contemporary sedimentation rates of rich in organic content matter (range in some regions even over 2 mm per year) caused by increased eutrophication [12]. All of these factors lead to the formation of free gas in sediments during the process of anaerobic decomposition of organic matter.

![Fig. 1. Location of the experiment](image-url)
3. METHODS

Active hydroacoustic methods were used in order to cover the largest possible bottom/area and to determine potential sites of the occurrence of gas saturated sediments. For this purpose, a number of research cruises were carried out from 2008 to 2014. During the research we have used several single beam echo-sounders with diverse frequency bands from 12 up to 200 kHz and echo-sounder emitted CW signals from 40 to 80 kHz to identify sea bottom gassy sediments, presence of pockmarks and for seeking gas seepages.

Shape of an echo envelope is directly related to the bottom physical properties. Parametric analysis of acoustic echo were performed basing on a volume backscattering strength (Sv) in order to determine range of occurrence of gaseous sediments and all types of inhomogeneities associated with the gas bubbles displacements between solid particles.

Main group of parameters used in classification process was statistical zero (n=0) order moment, second (n=2) order moment and third (n=3) order moment characterizing the variability of the shape of the echo envelope was calculated:

\[ \Lambda_{X_n} = 2 \int_0^\infty t^n X(t) dt \]  

Where t expressed reaching time of the selected echo fragment from bottom to the transmitter, and X(t) represents transformed echo envelope.

Basing on the second and third order moment we have calculate skewness of the echo envelope, using the equation:

\[ X_{sk} = \frac{\Lambda_{X_3}}{\sqrt[3]{\Lambda_{X_2}}} \]  

Second group of factors was wavelet parameters. Basis on the wavelet transform coefficient, wavelet decomposition energy parameters are calculated according to the equation:

\[ E_{\text{wav},j} = \int_0^b C(a,b) db \]  

Where \(C(a,b)\) is the wavelet transform coefficient (a is the scale factor and b corresponds to the shift of the wavelet) describing degree of correlation between wavelet and analysed part of the signal. In the next step, basis on the wavelet energy parameter, entropy of wavelet signal decomposition was calculated:

\[ \text{ent}_{\text{wav}} = \sum_{j=1} E_{\text{wav},j} \ln E_{\text{pwave}} \]  

Calculated parameters were used as input data for the principal component analysis and for the clustering algorithms: k-means method and based on the principles of fuzzy logic c-means method.

4. RESULTS AND DISCUSSION

Acoustic properties of sediments are largely dependent on the presence of gas bubbles in the bottom. Parameterization of the acoustic signals allowed to observe significant differences between echo scattered from the sediments with gas bubbles (in the pockmark area) and sediments surrounded pockmark (without gas bubbles). Calculated for the purposes
of the experiment group of statistical parameters shows high sensitivity to changes in sediment (Fig. 2.).

![Echogram of the gas pocket area](image)

Fig. 2. (A) Echogram of the gas pocket area (registered using 200 kHz echosounder). On the (B) panel, variabilities of the six first wavelet decomposition energy parameters are shown. On the panel (C) variabilities of the echo envelope parameters (1- wavelet signal decomposition entropy, 2-zero order moment, 3- second order moment, 4- skewness) are presented. On the panel (D) the results of the classification calculated applying k-means clustering method are shown.

Also wavelet decomposition energy parameters show significant differences between absolute values calculated for the areas with and without gas bubbles in the sediments. Clustering methods applied to this parameters enable to precise identification of the gaseous sediment zones.

Results obtained basis on the principle components (calculated using all parameters) allows for accurate distinction zones of gas occurrence in the bottom (Fig.3). Satisfactory results of the k-mean clustering can be obtained basing only on the first three components describing more than 85 percent of the variation in the envelope of the acoustic signal. The use of an c-mean algorithm basing on the assumptions of fuzzy logic enables to extract not only areas of the gas saturated sediments and sediments without gas, but also regions where there are sediment structures connected with migration of the gas bubbles such as gaseous chimneys under the pockmark crater (Fig. 4)

5. SUMMARY

Parametric Analysis of the echo envelope clearly demonstrates that presence of the significant amount of gas bubbles in sediments is one of main factors that have an influence on acoustic properties of the sea bottom. Described in this publication parameterization method of the acoustic echo envelope show high sensitivity for a changes in the signal associated with the occurrence of gas in the sediments. Using the clustering algorithms allows
for a very accurate identification both the gas saturated sediments as well as a sediment structures associated with the movement of the gas bubbles.

Fig. 3. (A) Echogram showing a fragment of the sediment without gas bubbles (left side) and fragment of the gaseous pockmark with visible outflow of gas bubbles (right side). (B) panel shows the results of the classification calculated applying k-means clustering method. Panels (C) and (D) present variability of the first seven principal components obtained on the basis of all calculated parameters.

Fig. 4. Echogram (obtained from 40 kHz echosounder) of the pockmark covered by thick layer of the mud. On the panel (A) results of the c-mean clustering (separation of two classes). On the panel (B) results of the c-mean clustering (separation of three classes).
ACKNOWLEDGEMENTS

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REFERENCES

COMPARISON OF THEORIES FOR ACOUSTIC WAVE PROPAGATION IN GASSY MARINE SEDIMENTS

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Abstract: More than three decades ago, Anderson and Hampton [1, 2] (A&H) presented theories for wave propagation in gassy water, saturated sediments and gassy sediments in their two-part review, which has been cited by many researchers in the geoacoustics and underwater acoustics areas. They gave an empirical formulation based on the theory of Spitzer [3] for the wave propagation in gassy water by adapting that for a viscoelastic, lossy medium. Following Leighton [4], this paper presents a theory based on non-stationary nonlinear dynamics of spherical gas bubbles and extends that 2007 paper to include liquid compressibility and thermal damping effects. The paper then shows how that nonlinear formulation can be reduced to the linear limit, and derives the expressions for the damping coefficients, the scattering cross section, the speed of sound and the attenuation, and compares these with the A&H theory. The current formulation has certain advantages over A&H theory such as implementing an energy conservation based nonlinear model for the gas pressure inside the bubble, having no sign ambiguity for the speed of sound formula (which is important when estimating the bubble void fraction) and correcting the ambiguity on the expression for scattering cross section, as identified in the recent work of Ainslie and Leighton [5]. Moreover, the theory presented here forms a basis for a nonlinear, time-dependent acoustic estimation model for gas bubble distributions in viscoelastic mediums since it avoids the commonly encountered assumptions on the bubble dynamics such as linearity, steady-state behaviour and monochromaticity.

Keywords: gassy sediments, nonlinear propagation, acoustic scattering
1. INTRODUCTION

The presence of gas and its effects on the physical properties of the marine sediment are of interest for several applications, including drilling operations, construction of seafloor structures, and environmental considerations such as global warming, climate change and the slope stability of the sediments. The use of acoustics in order to characterize the physical properties of ocean bottom sediments is of increasing interest. This has highlighted the need for an in-depth and accurate theoretical framework for acoustic wave propagation in gasy mediums. Currently, the theory used by the majority of investigators relies mostly on the benchmark work of Anderson and Hampton [1, 2] (abbreviated as A&H theory hereafter) which was, at the time, an extensive review of the theories for wave propagation in gasy water, saturated sediments and gasy sediments. They provide an empirical formulation based on the theory of Spitzer [3].

The A&H theory needs to be reconsidered owing to following reasons: (i) it assumes that only linear, steady state pulsations occur, which makes the method inapplicable for high amplitude pulses and second harmonic or combination frequency signals [4]; (ii) the expression for the viscoelastic losses are given \textit{a posteriori} without a rigorous derivation, leading to some ambiguities [5-7]; (iii) at a later date, Prosperetti et al. [8] presented a formulation for the thermal behaviour of the gas pressure inside the bubble, which is more complete than the use of polytrophic relation A&H employed (especially when bubble resonance effects are present) and can be incorporated into the current problem; and (iv) the expression for the scattering cross-section, when used together with the radiation damping, involves an inconsistency in terms of frequency dependence of the expressions [5].

Leighton [4] presented a theory based the non-stationary nonlinear dynamics of gas bubbles in marine sediments, noting the requirement for follow-on work to include liquid compressibility and thermal damping effects. This paper undertakes that follow-on work, and then in the linear limit, derives expressions for the damping coefficients, the scattering cross-section, the speed of sound and the attenuation.

2. THEORY

Following Yang and Church [9], the Keller-Miksis type equation which describes the radial motion of a spherical bubble in an unbounded viscoelastic medium can be written as follows;

\[
\left(1 - \frac{\dot{R}}{c}\right) \ddot{R} + \frac{3}{2} \left(1 - \frac{\dot{R}}{3c}\right) \dot{R}^2 = \left(1 + \frac{\dot{R}}{c}\right) \frac{p_L - p_\infty}{\rho} + \frac{R}{\rho c} \frac{d}{dt}(p_L - p_\infty)
\]  

(1)

where

\[
p_L - p_\infty = p_R = \frac{2\sigma}{R} - p_0 + p_A g(t) - \left[\frac{4G}{3R^3} (R^3 - R_0^3) + \frac{4\mu R}{R}\right].
\]  

(2)
In equations (1) and (2), $R$ is the bubble radius, the dots indicate the time derivatives, $c$ is the speed of sound in the host medium, $\rho$ is the density of the medium, $p_i$ is the pressure outside the bubble wall, $p_\infty$ is the pressure at infinity, $p_g$ is the pressure inside the bubble, $\sigma$ is the surface tension, $P_A g(t)$ is the time-dependent acoustic pulse with $P_A$ being a positive real number that scales the driving pressure, $p_0$ is the static pressure, $G$ is the shear modulus and $\mu$ is the shear viscosity of the surrounding medium.

The continuity and the energy conservation equations for a perfect gas are given respectively by

$$\frac{D \rho_g}{Dt} + \rho_g \nabla \cdot \mathbf{v}_g = 0 \quad (3)$$

$$\rho_g C_p \frac{DT}{Dt} + \frac{T \partial \rho_g}{\rho \partial t} \left| \frac{Dp_g}{Dt} \right| = \nabla \cdot (K \nabla T). \quad (4)$$

In above equations, $\rho_g$ is the density of the gas, $\mathbf{v}_g$ is the velocity field within the bubble, $T$ is the temperature, $C_p$ is the specific heat at constant pressure, and $K$ is the thermal conductivity of the gas. The above formulation can be modelled to first order accuracy by using an artificial thermal viscosity term $\mu_{th}$ defined in [8].

An analytical solution to (1) may be obtained by assuming small perturbations of the bubble radius, i.e. $R = R_0 (1 + x(t))$ where $x << 1$:

$$\ddot{x} + 2 \beta_{tot} \dot{x} + \omega_0^2 x = -\frac{P_A e^{i\omega t}}{m} \quad (5)$$

where $m = \rho R_0^2 + 4(\mu_{th} + \mu) R_0/c$ is the effective mass, $\beta_{tot}$ is the total damping and $\omega_0$ is the natural frequency. The expressions for the viscous, thermal (using thermal viscosity), acoustic, interfacial and elastic damping obtained in this way are given respectively as

$$\beta_{vis} = 2\mu / \left( \rho R_0^2 + \frac{4(\mu_{th} + \mu) R_0}{c} \right), \quad (6a)$$

$$\beta_{p,th} = 2\mu_{th} / \left( \rho R_0^2 + \frac{4(\mu_{th} + \mu) R_0}{c} \right), \quad (6b)$$

$$\beta_{ac} = \frac{(\omega R_0/c)}{1 + (\omega R_0/c)^2} \frac{\omega}{2} \frac{(\rho R_0^2)}{\left( \rho R_0^2 + \frac{4(\mu_{th} + \mu) R_0}{c} \right)}, \quad (6c)$$

$$\beta_{int} = -\sigma / (\rho c R_0^2 + 4(\mu_{th} + \mu) R_0), \quad (6d)$$

$$\beta_{el} = 2G / (\rho c R_0 + 4(\mu_{th} + \mu)), \quad (6e)$$

where

$$\beta_{tot} = \beta_{vis} + \beta_{p,th} + \beta_{ac} + \beta_{int} + \beta_{el}, \quad (6f)$$

and the natural frequency satisfies
\[ \omega_0^2 = \left[ 3 \kappa \rho \delta_\theta - \frac{2 \sigma}{R_0^4} + 4G + \frac{\omega^2 \rho R_0^2}{1 + (\omega R_0/c)^2} \right]/m. \]  

(7)

The expressions for the non-dimensional thermal \((\delta_{\text{th}})\), elastic \((\delta_{\text{el}})\) and acoustic \((\delta_{\text{ac}})\) damping coefficients in A&H theory are given in Eq. (8), (9) and (10), respectively, of [2].

**Scattering cross-section**

Ainslie and Leighton [5] derived the equation for scattering cross-section of gas bubbles in water as

\[ \sigma_s = \frac{4\pi R_0^2}{\left( \frac{\omega_0^2}{\omega^2} - 1 - 2 \frac{\beta_0}{\omega} \varepsilon \right)^2 + \left( 2 \frac{\beta_0}{\omega} + \omega_0^2 \varepsilon \right)^2}. \]  

(8)

where \(\varepsilon \equiv \omega R_0/c\) and \(\beta_0\) is the total of the damping coefficients other than acoustic damping, this is also applicable to gassy sediments provided that correct expressions for \(\beta_0\) and \(\omega_0\) are used.

**Speed of sound and attenuation**

The complex speed of sound in a bubbly mixture, \(c_m\), is given by the expression [10, 11]

\[ \frac{c^2}{c_m^2} = 1 + 4\pi c^2 \int_0^\infty \frac{R_0 n(R_0)}{\omega_0^2 - \omega^2 + 2i\beta_{\text{tot}} \omega} \, dR_0. \]  

(9)

where \(n(R_0)\,dR_0\) is the number of bubbles per unit volume with radii between \(R_0\) and \(R_0 + dR_0\). Note that \(\beta_{\text{tot}}\) includes the elastic damping and the interfacial damping in addition to the other damping mechanisms for bubbles in water. Setting \(c/c_m = u - iv\) yields expressions for phase velocity \(V\) [10]

\[ V = c / u, \]  

(10)

and attenuation \(A\) in dB/cm

\[ A = 8.6859 (\omega u/c). \]  

(11)

3. RESULTS

**Application of the model to sediments**

The proposed model can be applied to sediments with several advantages over A&H model such as having no sign ambiguities in the speed of sound formula and defining the higher order scattering coefficients. In this section, the results obtained by applying the model to the marine sediments are presented and compared to those obtained by A&H model. The formulation of the A&H model is not explicitly stated in this paper. Two different sediments types, ocean bottom silt and harbour mud, which were investigated by A&H will be examined here as
well, to facilitate a direct comparison of two models. The mixture properties of ocean silt and harbour mud, as given in A&H, are repeated here in Table 1. Air bubbles (with polytrophic exponent $\kappa = 1.4$) are assumed to be embedded in sediments. Density of seawater is taken as $1030 \text{ kg/m}^3$ based on a salinity of 0.3%.

<table>
<thead>
<tr>
<th></th>
<th>Harbour Mud</th>
<th>Ocean Silt</th>
</tr>
</thead>
<tbody>
<tr>
<td>Porosity</td>
<td>0.75</td>
<td>0.68</td>
</tr>
<tr>
<td>Shear Modulus ($G$)</td>
<td>1 GPa</td>
<td>250 GPa</td>
</tr>
<tr>
<td>Bubble void fraction ($\varepsilon$)</td>
<td>0.075</td>
<td>0.068</td>
</tr>
<tr>
<td>Density ($\rho$)</td>
<td>1400 kg/m$^3$</td>
<td>1550 kg/m$^3$</td>
</tr>
<tr>
<td>Speed of sound ($c$)</td>
<td>1488 m/s</td>
<td>1552 m/s</td>
</tr>
</tbody>
</table>

Table 1: Model input parameters for harbour mud and ocean silt

### Linear damping coefficients

In this section, damping constants of the current formulation are plotted for air bubbles in ocean sediments and compared to the predictions from the A&H theory by assuming $\delta \equiv 2\beta/\omega$, where $\delta$ is the non-dimensional and $\beta$ is the dimensional damping coefficient.

In Fig. 1a and 1b, the linear damping coefficients for acoustic propagation in harbour mud are plotted as a function of frequency using Eq. (6) and A&H theory, respectively, and in silt, they are plotted in Fig. 1c and 1d, respectively. First of all, the two formulations show identical results for the acoustic damping. As Ainslie and Leighton [5, 7] recently noted, there existed a contradiction for many years for the expression of acoustic damping, A&H being among the few who have reported this issue and used the correct expression. For the thermal damping, the two formulations predict quite different results, especially in terms of the trend they show with increasing applied driving frequency. This is mainly due to the thermal models used. This paper uses a nonlinear, energy conservation formulation given in Eqns. (6) and (7) for the gas inside bubble whereas A&H modifies Devin [12]’s approach which solves energy conduction equation assuming variable gas stiffness which can take values between the isothermal and adiabatic, and intermediate stages.

It is interesting to note that the two formulations predict slightly different value for the applied frequency at which the minimum damping is observed (also where the maximum bubble resonance is expected). For instance, the resonance frequency of a 1 mm bubble in mud is $\sim 8.5$ kHz according to current theory and it is $\sim 8.9$ kHz according to A&H. Considering the fact that the use of thermal viscosity is not involved in the expression for the resonance frequency (thus neglecting such thermal effects on resonant frequency) the current theory predicts a cross-over of elastic and acoustic damping at around that frequency. The minimum damping in A&H occurs at a slightly higher frequency. This discrepancy can be explained as follows: A&H use complex dynamic shear modulus $G^* = G + iG'$ where the imaginary part $G'$ is taken a priori as $G/5$. The expression for the resonance frequency involves the real part of $G^*$ whereas the viscoelastic losses are evaluated by introducing a term analogous to viscosity,
\( \mu_f \equiv G' / \omega \). The latter fact also explains the reason why the viscoelastic damping in Figs. 1b and 1d decreases with increasing driving frequency. The resonance frequency for a 1 mm bubble in silt is predicted \(-127\) kHz from both theories, which is slightly underestimated by A&H theory according to the minimum damping observed in Fig. 1d.

Viscous and interfacial damping values of current theory are not shown in the figures because they have much lower values compared to other sources of damping. However, the curves for \( \beta_{tot} \) include also those values in Fig. 1a and 1c.

Among all damping mechanisms, the elastic damping is apparently the most important. The two formulations use slightly different constitutive models which is the reason for the observed differences in elastic damping. However this observation is currently under further investigation.

---

**Fig. 1:** Damping coefficients vs driving frequency for a 1 mm equilibrium radius spherical bubble in mud using (a) Eq. (6) and (b) the A&H theory with dimensional coefficients (\( \delta \omega / 2 \)), and in ocean silt using (c) Eq. (6) and (d) the A&H theory with dimensional coefficients (\( \delta \omega / 2 \)). Viscous and interfacial damping coefficients are not plotted for clarity though \( \beta_{tot} \) includes them as in Eq. (6f).

**Scattering cross section**
Calculated values of scattering cross-section of bubbles in harbour mud and ocean silt, normalized with respect to their geometrical cross sections, are plotted in Fig. 2a and 2b, respectively. The current theory predicts a cross section almost two orders of magnitude higher for a bubble with radius approximately 1 mm in mud, whereas it predicts a smaller value for near resonant bubbles in ocean silt.

![Figure 2: Scattering cross-sections for bubbles in (a) harbour mud and (b) in ocean silt, driven by a 10 kHz pulse, calculated using Eq. (8) and A&H theory.](image)

**Speed of sound**

Computed values of speed of sound are plotted in Fig. 3 for bubbles in harbour mud for two different values of bubble void fraction specifically 0.075% and 7.5%. It is observed that for wave propagation in harbour mud the current formulation estimates reduced damping in the frequency range where bubble resonance effects are more significant. This is consistent with the results in Fig. 1a and 1b where the total damping, in the range 1-100 kHz, for a 1 mm bubble is predicted to be less than that in A&H.

![Figure 3: Speed of sound through a mono-disperse bubble population in harbour mud plotted using Eq. (10) and the A&H theory for bubble void fraction values of 0.075% and 7.5%.](image)
In Fig. 4, the wave propagation through a mono-disperse bubble cloud in ocean silt is plotted for the gas void fraction values 0.068% and 6.8%. One may notice that the below resonance and the above resonance speed of sound values are similar to each other and that the transition near the resonance regime is smooth.

![Figure 4: Speed of sound through a mono-disperse bubble population in ocean silt plotted using Eq. (10) and the A&H theory for bubble void fraction values of 0.068% and 6.8%. The results from both formulations for $\Gamma = 0.068\%$ are very similar to each other such that the two curves overlie one another.](image)

**Attenuation**

Computed values of attenuation in gassy sediments are plotted in Fig. 5a and Fig. 5b for bubbles in harbour mud and ocean silt, respectively. The values are computed using the same values as in the previous plots for speed of sound, i.e. using $\Gamma = 7.5\%$ for harbour mud and $\Gamma = 6.8\%$ for ocean silt. The solid black lines in Fig. 5 give the result obtained by A&H (their Fig. 16 of Part II). It is observed that the current formulation predicts lower attenuation in harbour mud for pulse frequency of less than 10 kHz. For ocean silt the predictions from the two formulations are quite similar, this is a fact consistent with the results obtained for the speed of sound.
Figure 5: Attenuation of acoustic wave through a mono-disperse bubble population (a) in harbour mud for $\Gamma = 7.5\%$ and (b) in ocean silt for $\Gamma = 6.8\%$, plotted using Eq. (11) and the A&H theory.

4. CONCLUSIONS

The new formulation proposed in this paper has removed the ambiguities with the scattering cross-section and speed of sound expressions from the A&H formulations, and is not restricted to linear monochromatic bubble pulsations. This is important because many methods for detecting bubbles rely on bubble behaviours which are not restricted to these limitations [12, 13]. However when reduced to the linear regime to allow comparison with the predictions of the A&H formulation, the new method shows considerable and important differences in damping, bubble resonance, sound speed and attenuation. These differences need to be checked to ascertain if the new predictions are correct, because errors in the well-used A&H formulation would be inherent (though it is too soon to say whether they are also important) in many studies that use of the A&H formulation.

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ATTENUATION OF LOW FREQUENCY UNDERWATER NOISE USING ARRAYS OF AIR-FILLED RESONATORS

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Abstract: This paper investigates the acoustic behavior of underwater air-filled resonators that could potentially be used in an underwater noise abatement system. The resonators are similar to Helmholtz resonators without a neck, consisting of underwater inverted air-filled cavities with combinations of rigid and elastic wall members, and they are intended to be fastened to a framework forming a stationary array surrounding a noise source, such as a pile driving operation, a natural resource production platform, or an air gun array. Previous work has demonstrated the potential of surrounding noise sources with arrays of large stationary encapsulated bubbles that can be designed to attenuate sound levels over any desired frequency band and with levels of reduction up to 50 dB [Lee and Wilson, Proceedings of Meeting on Acoustics \textbf{19}, 075048 (2013)]. Open water measurements of underwater sound attenuation using resonators were obtained during a set of lake experiments, where a low-frequency electromechanical sound source was surrounded by different arrays of resonators. The results indicate that air-filled resonators are a potential alternative to using encapsulated bubbles for low frequency underwater noise mitigation.

Keywords: underwater noise, anthropogenic noise, noise abatement
1. INTRODUCTION

The goal of this work is to investigate the efficacy of using air-filled, open-ended resonators to abate low frequency underwater noise and to compare their performance with large encapsulated bubbles. Previous work demonstrated the use of arrays of large tethered encapsulated bubbles to attenuate underwater sound in the 50 Hz to 1000 Hz frequency band from a variety of continuous and impulsive sources. [1–5] Air-filled encapsulated bubbles can be tethered to a framework surrounding a noise source, such as a pile driving operation or a natural resource production platform, to reduce radiated sound in a frequency band coincident with the peak sound levels emitted by the source. The volume of each encapsulated bubble in the array is designed so that its acoustic resonance occurs at a frequency near the low end of a chosen source’s noise spectrum. Acoustic energy is damped near and above the encapsulated bubble resonance frequency primarily via phase incoherent acoustic re-radiation for the large bubbles sizes (approximately 10 cm in diameter) used in this application—energy from the sound wave goes into oscillating the bubbles in the array. While viscous and thermal damping also occur, they play a much smaller role for such large bubble sizes. Encapsulated bubbles that have a higher $Q$ (quality factor) at resonance are better oscillators and can provide more sound level reduction from this radiation damping. The typical embodiment of an encapsulated bubble used in a noise abatement system is an air-filed rubber balloon. The encapsulating rubber shell is typically thick enough (~1 mm–2 mm) that it can survive deployment and withstand the marine environment.

As an alternative to encapsulated bubbles, a potentially more effective type of resonator was developed, consisting of a container with a combination of rigid and elastic wall members and a single open end. The container is inverted so that the open end at the bottom forms an air-water interface, similar in concept to an air-filled underwater Helmholtz resonator without a neck, such that the air volume can undergo driven oscillations and sound can be re-radiated from the air-water interface at the opening and at the rubber sides. [6] The new resonator had rectangular box geometry with an open end on the bottom and a total enclosed volume of 216.8 cm$^3$. Instead of making all of the walls rigid, two vertical walls of new container consisted of thin rubber sheets (less than 1 mm thickness) to increase the amount of radiating surface area.

These new open-ended resonators were fabricated and tested in a series of laboratory and open-water experiments. The laboratory tank experiments were conducted to measure the resonance frequency and quality factors of both individual open-ended resonators and encapsulated bubbles. The new open-ended resonator was designed so that it had approximately the same resonance frequency as a selected size of encapsulated bubble (~110 Hz at an encapsulated bubble volume of 2671.5 cm$^3$), although the open-ended resonator had nearly double the quality factor $Q$ of the encapsulated bubble. An additional feature of the new open-ended resonator is that it exhibits a second additional resonance below the primary resonance frequency. Attenuation measurements performed in a fresh water lake demonstrated that for a fixed void fraction, the array of open-ended resonators provided at least three times the amount of peak attenuation compared to an encapsulated bubble array of the same void fraction.

2. LABORATORY RESONANCE FREQUENCY AND QUALITY FACTOR MEASUREMENTS
Laboratory measurements of resonance frequencies and quality factors of individual open-ended resonators and encapsulated bubbles were made using the closed, water-filled tank apparatus and data analysis techniques described in Ref. 7. An individual open-ended resonator or encapsulated bubble was placed inside the tank apparatus, and a piston driven by an electromechanical shaker excited the acoustic response of the open-ended resonator or encapsulated bubble and of the tank. The pressure radiated by the resonator under test was measured by a hydrophone mounted inside the tank, and a spectral subtraction technique was used to estimate its frequency response. [7,8]

The resonance peaks for each of the resonator types are shown in Fig. 1. The resonance frequency $f_0$ corresponds to the maximum amplitude and the quality factor at resonance is given by $Q = f_0 / \Delta f$, where $\Delta f$ is the width of the peak 3.01 dB down from the maximum amplitude. A single peak is observed in the resonance spectrum for the encapsulated bubble; however, two peaks occur in the open-ended resonator’s response. The measured resonance frequencies and quality factors are listed in Table 1.

![Resonance spectra of an individual open-ended resonator and an encapsulated bubble measured in a laboratory tank. Both types of resonators display resonance peak between 113 Hz and 116 Hz although the open-ended resonator design has a secondary lower frequency resonance at 83.6 Hz.](image)

<table>
<thead>
<tr>
<th>Resonator type</th>
<th>Primary resonance frequency [Hz]</th>
<th>$Q$ at primary resonance</th>
<th>Secondary resonance frequency [Hz]</th>
<th>$Q$ at secondary resonance</th>
</tr>
</thead>
<tbody>
<tr>
<td>Open-ended</td>
<td>113.2</td>
<td>6.0</td>
<td>83.6</td>
<td>5.1</td>
</tr>
<tr>
<td>Encapsulated</td>
<td>115.7</td>
<td>3.6</td>
<td>–</td>
<td>–</td>
</tr>
</tbody>
</table>

Table 1: Resonance frequencies and quality factors for open-ended resonators and encapsulated bubbles.

The encapsulated bubble consisted of a rubber-shelled, air-filled balloon with a volume of 2671.5 cm$^3$. Such encapsulated bubbles have been the subjects of previous investigations, and their acoustic behavior is well predicted by Church’s model. [4,7,9]
The open-ended resonator has a primary resonance (highest amplitude resonance) at nearly the same frequency as the encapsulated bubble, but with a much smaller volume of 216.8 cm$^3$—about 1/12th the volume of the encapsulated bubble. Compared to the encapsulated bubble, the open-ended resonator has a measured $Q$ that is greater by a factor of 1.67. Additionally, the open-ended resonator exhibits a secondary, lower frequency resonance at 83.6 Hz with a similarly high $Q$. Although a detailed explanation of this remains the subject of further investigation, a potential reason for the discrepancy in acoustic behavior and between air volumes could be that while the air volume of the encapsulated bubble is fully constrained by its rubber shell, the air volume of the open-ended resonator is partially constrained by both a rigid member and rubber sheets in addition to having one unconstrained surface. The combined effects of the various impedances at the different boundaries of the open-ended resonator’s enclosed air volume likely contribute to its unique acoustic properties.

3. OPEN-WATER ATTENUATION MEASUREMENTS

Attenuation measurements using arrays of both resonator types were conducted from an anchored, floating barge located in Lake Travis, a fresh water lake near Austin, Texas. Collections of either encapsulated bubbles or open-ended resonators were attached to netting stretched across the outer sides of a steel frame with height of 1.3 m and horizontal dimensions of 1.2 m by 1.2 m. Two additional vertical panels of netting were attached to the interior of the frame so that the resonators formed a three-dimensional volumetric array. Two steel weights were attached to the rigid framework to provide additional ballast. A US Navy J-13 low frequency reference projector was suspended in the middle of the resonator array, and the entire apparatus was submerged into the lake using an overhead gantry crane such that the center of the frame was at a depth of 1.2 m. A pre-deployment photograph of the apparatus with 96 open-ended resonators attached to it is shown in Fig. 2a. A High Tech, Inc. HTI-90 hydrophone was lowered into the water 11.5 m away from the resonator array. The acoustic pressure was measured in increments of 2 m, from 2 m to 20 m depth with and without the various resonator arrays present. The lake depths beneath the resonator array and hydrophone deployment sites were 20.9 m and 21.2 m, respectively. A schematic of the experiment configuration is shown in Fig. 2b.

Various numbers of each resonator type were attached to the test framework to achieve different void fractions. The void fraction is given by the expression $\beta = N V_{res}/V_{tot}$, where $N$ is the number of resonators, $V_{res}$ is the volume of an individual resonator, and $V_{tot}$ is the total volume contained within the test framework. Void fractions of 0.08%, 0.17%, 0.34%, 0.67%, and 1.01% were obtained using 8, 16, 32, 64, and 96 open-ended resonators. Using 8, 16, and 32 of the larger volume encapsulated bubbles, void fractions of 1.03%, 2.07%, and 4.14% were achieved.

Narrowband spectra (4096 FFT points, 0.488 Hz resolution bandwidth) of the measured acoustic pressure for each case are plotted in Fig. 3. The spectra are normalized such that highest pressure in the baseline case (the iteration with no open-ended resonators or encapsulated bubbles present) is equal to 0 dB. As the number of resonators of either type increases, and hence the void fraction, is increased, the overall sound level received at the hydrophone is reduced. For the highest void fraction cases, the sound level reduction near the individual resonator resonance frequency is great enough that the signal level is coincident with the ambient noise floor. Sharp lines occurring in the spectra at 60 Hz and higher harmonics are due to acoustic noise radiated by vibrating electrical transformers attached to the deck of the test barge, hence this noise is not attenuated by the resonator.
array since its source is located outside of the array. Increased spectral levels occurring below the individual resonator resonance frequency are due to enhanced coupling of the source signal into the lake waveguide through collective in-phase oscillations of the resonator array.

![Image of resonator array](image1)

**Fig. 2:** (a) Photograph of open-ended resonator array with 96 resonators attached to test framework prior to being deployed. The J-13 sound source was suspended in the center of the array. (b) Schematic of the sound source and receiver configuration used in the lake experiment.

![Schematic of sound source and receiver configuration](image2)

**Fig. 3:** Example measured acoustic pressure from a hydrophone 11.5 m away and at a depth of 10 m for the various configurations of open-ended resonators and encapsulated bubbles. Left: Open-ended resonators arrays—$N = 8$, $\beta = 0.08\%$ (light blue), $N = 16$, $\beta = 0.17\%$ (magenta), $N = 32$, $\beta = 0.34\%$ (green), $N = 64$, $\beta = 0.67\%$ (dark blue), and $N = 96$, $\beta = 1.01\%$ (red). Right: Encapsulated bubble arrays—$N = 8$, $\beta = 1.03\%$ (red), $N = 16$, $\beta = 2.07\%$ (dark blue), and $N = 32$, $\beta = 4.14\%$ (green). The black solid lines in both plots correspond to the baseline spectrum, and the grey dashed lines show the ambient noise spectrum. The red solid lines in both plots correspond to the arrays with approximately 1% void fraction.

The narrowband spectra for each resonator or baseline case were divided into 25-Hz-
width frequency bins, and the mean amplitude was computed for each bin. These band levels for each case were then averaged over each of the receiver depth positions. To estimate the band level reduction for each resonator array, the band levels for a particular case were subtracted from the baseline band levels. The resultant depth-averaged band level reductions are plotted in Fig. 4 for three different cases: 1.01% void fraction array of open-ended resonators, 1.03% void fraction array of encapsulated bubbles, and 4.14% void fraction array of encapsulated bubbles. For a fixed void fraction of ~1%, the open-ended resonators provided at least three times the peak reduction in the band closest to the resonance frequencies of the two types of resonators centered at 114 Hz. The increased attenuation of the source signal is both due to the higher $Q$ of the open-ended resonators and also likely to an increased impedance mismatch around the source from the higher number of air-filled resonators. The band level reduction provided by the 4.14% void fraction encapsulated bubble array is comparable to the 1.01% void fraction open-ended resonator array for frequencies up to about 1 kHz. For higher frequencies the reduction from the encapsulated bubble array falls off to zero while the open-ended resonator continues to provide at least 10 dB of reduction up to approximately 1.8 kHz. As the wavelength approaches the array size at these higher frequencies, scattering effects may become important, and the comparatively larger number of the smaller open-ended resonators per wavelength may enhance the scattering effects. Finally, the sub-resonance band level increase is more pronounced with the open-ended resonator array than in the encapsulated bubble cases. This effect is also likely enhanced by the greater number of open-ended resonators in the array, which could potentially increase the efficiency of the in-phase collective oscillations of the array below the individual resonator resonance frequency.

**Fig. 4:** Depth-averaged band level reduction for three different array cases: 96 open-ended resonators, $\beta=1.01\%$, 8 encapsulated bubbles, $\beta=1.03\%$, and 32 encapsulated bubbles, $\beta=4.14\%$. The frequency bands each had a width of 25 Hz.

Because ballast weight is required to hold down both the open-ended resonators and the encapsulated bubbles, there is an advantage to having a given level of reduction
achieved through a reduced void fraction array of open-ended resonators. Approximately four times less ballast is required to submerge the 1.01% void fraction open-ended resonator array than the 4.14% encapsulated bubble array. The broadband level reductions for the 100 Hz to 1 kHz frequency range and each of the open-ended resonator and encapsulated bubble cases are plotted versus the amount of ballast needed to just make each array negatively buoyant, shown in Fig. 5. The 4.14% void fraction encapsulated bubbles array requires nearly an order of magnitude more ballast to attain negative buoyancy compared to the similarly performing 1.01% void fraction open-ended resonator array. In practice, noise abatement systems made from these open-ended resonators would have reduced manufacturing, shipping and deployment costs compared to equivalent-performance encapsulated bubble systems because of the reduced need for ballast.

![Fig. 5: Mean sound level reduction in the 100 Hz to 1000 Hz band plotted versus the minimum ballast mass needed to submerge each array. On such a plot, systems with curves shifted to the left are lighter, and less expensive to build, ship and deploy.](image)

4. SUMMARY

A new prototype open-ended resonator design was developed and tested for the purpose of incorporating arrays of the resonators into an underwater noise abatement system. Individual resonators were designed to have a resonance frequency near 100 Hz, similar to encapsulated bubbles that were used in previous noise abatement systems. Laboratory measurements confirmed that open-ended resonator had a resonance frequency at 113.2 Hz compared to the resonance frequency of the encapsulated bubbles used in the testing with resonance frequencies of 115.7 Hz, but at only one-twelfth the air volume of a single encapsulated bubble. Furthermore, the new resonator possesses a second resonance frequency at 83.6 Hz not exhibited by the encapsulated bubble. The $Q$ values for both resonances of the open-ended resonator are 1.5–2 times larger than that of the encapsulated bubble, indicating that the new resonator is a comparatively better oscillator. Attenuation measurements conducted in a lake confirmed that the an array of open-ended resonators can provide at least three times the amount of peak sound level reduction than an encapsulated bubble array of the same void fraction. Additionally, an open-ended
resonator-based noise abatement system has the potential to achieve a target amount of underwater sound attenuation, but at four times less required ballast weight than an equivalently performing encapsulated-bubble-based system due to the increased acoustic efficiency of the open-ended resonator design.

5. ACKNOWLEDGEMENTS

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REFERENCES

NUMERICAL MODELLING OF A BUBBLE CURTAIN

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Abstract: The offshore wind energy plays an important role in the energy concept of the German Federal Government. The pile driving during the construction of the foundation causes high sound pressure levels in the sea. This noise emission can be potentially dangerous for the marine life, especially mammals. The application of different sound mitigation systems, for example big bubble curtains, promises a reduction of the pressure level. The lack of information about the physical effects makes it difficult to evaluate these systems and their potential for optimization.

In this paper a method based on the wave equation for modelling a bubble curtain in detail is presented. First, the method is used to analyse a simple bubble curtain with equidistant bubbles of the same size. In the next step the interaction between two bubbles is considered.

First results show that the bubble oscillation and the interaction between bubbles have to be taken into account for the typical excitation wavelength associated to pile driving.

Keywords: bubble curtain, bubble oscillation, bubble cluster, wave equation, numerical modelling
1. INTRODUCTION

To protect the marine environment, sound pressure levels have to be considered during offshore pile driving. To avoid permitted limits to be exceeded, noise reduction systems must be used during the construction. The bubble curtain is a commonly used system to reduce the acoustic pressure. Air is pumped through a tube laying on the ground or hanging inside the water. It leaves the tube through defined holes and rises due to buoyancy in form of bubbles. The bubble curtain can be placed directly at the pile or at a certain distance.

The validation of the sound pressure reduction due to a bubble curtain located in the sound propagation path is only possible by direct measurements. A comprehensive sound propagation model for piling noise up to a range of 5 km is being developed in cooperation with the Hamburg University of Technology and the University of Kiel as part of the Project BORA, funded by the Federal Ministry for Economic Affairs and Energy. The model should also take into account existing noise reduction systems.

The reasons for the sound attenuation effect of the bubble curtain are not clearly understood. This is partly due to the fact that direct measurements of the properties of the bubble curtain, such as the bubble size distribution, are very difficult to obtain because of the stochastic nature of the bubble flow and the uncertain boundary conditions in the sea. Rustemeier et al. [1] estimated the bubble size distribution under laboratory conditions in a basin and carried out a large acoustic sensitivity study with a bubble curtain in a test lake. They observed that small bubbles generated by a perforated coating have significantly higher mitigation effects than bubbles generated by a normal tube. In addition they compared the experimental data with a model based on the work described by Novarini et al. in [2]. This model considers the water-bubble mixture as an equivalent fluid with an effective sound speed and density. Next to the eigenfrequency of the bubble the model predicts a rising of the compressibility. Interactions between bubbles are not considered.

Accordance with the measurements could only be achieved by assuming much bigger bubbles than those observed. Due to the little information available about bubble curtains, this paper presents a detailed model covering the different physical effects involved in their insulating effect. The model, based on the wave equation, considers the water and air phases separately. In the first step the detailed model is described. Afterwards the results obtained for a simple bubble curtain with equidistant bubbles of the same size are presented. Then, the effect of the interaction between bubbles is analysed for a simple case with two identical bubbles. Finally the main conclusions and future work are summarised.

2. DETAILED MODEL OF THE BUBBLE CURTAIN

To model the noise mitigation properties of a bubble curtain, especially those regarding to the oscillation of the bubbles, the air and water phase have to be considered separately. Due to the size of the bubble curtain and the amount of bubbles involved, the computational cost of modelling the whole bubble curtain is unaffordable. An alternative approach is to consider a section of the curtain. The stochastic nature of the bubble curtain requires the considered control volume to be statistically equivalent to the whole curtain, for instance regarding to the bubble size distribution.
In Fig. 1 the bubble curtain and the observed control volume with the corresponding boundary conditions are plotted. The bubble is located in the middle of the volume. The plane pressure wave is entering the domain at the left. On the right side a non reflecting boundary condition is defined. On the remaining faces symmetry conditions are applied. The resulting pressure field is obtained by solving the wave equation in frequency domain (i.e. Helmholtz equation) with the finite element method.

3. RESULTS

In a first step a simple case of a bubble curtain consisting of equidistant bubbles of the same size is considered to validate the model. The distance among the bubbles is 40 times the bubble radius, so a low interaction between the bubbles can be assumed. The air volume fraction is 0.13% and an incoming pressure of 10 000 Pa is applied at each Hz.

Fig. 2a shows the resulting pressure in the bubble for the whole frequency range of interest and three different bubble radii. As reported by Medwin and Clay [3], the bubble behaves as a mass-spring system next to the first bubble eigenfrequency, leading to high values of the pressure. Tab. 1 compares the determined eigenfrequencies with the Minnaert frequency (or mass-spring resonance frequency) defined in [4]. The eigenfrequencies calculated by the model differ about 4% of those determined by the literature. The conclusion is that the bubble oscillation can be captured by the wave equation.

Fig. 2: a) Absolute pressure in the bubble at different frequencies for different radii. b) Absolute pressure averaged over the right side of the control volume
The averaged pressure over the right side of the control volume is computed to determine the noise mitigation properties of the bubble curtain. In Fig. 2b this value is plotted for the same three bubble radii of Fig. 2a. Next to the eigenfrequencies of the bubbles a reduction up to 20dB can be observed. For frequencies larger than these values the sound pressure rises up to the value of the incident pressure wave.

<table>
<thead>
<tr>
<th>Bubble radius [mm]</th>
<th>Eigenfrequency [Hz]</th>
<th>Minnaert frequency [Hz]</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>680</td>
<td>653</td>
</tr>
<tr>
<td>2</td>
<td>1696</td>
<td>1632</td>
</tr>
<tr>
<td>5</td>
<td>3401</td>
<td>3265</td>
</tr>
</tbody>
</table>

*Tab. 1: Comparison of the bubble eigenfrequency with Minnaert Frequency*

The fact that the sound reduction occurs only at the natural frequency of the bubbles leads to the conclusion that the dominant phenomenon is the scattering resulting from the oscillation of the bubble. In Fig. 3, the distribution of the absolute pressure in a section of the control volume is shown for the case of resonance of the bubble. Due to the high pressure amplitude in the bubble, the bubble itself becomes a source of a spherical wave.

The resulting pressure field can be interpreted as a superposition of the incoming pressure field and the field scattered by the bubble.

In the next step a bubble cluster consisting of two bubbles of the same size (radius: 1 mm) is considered. The cluster is perpendicular to the incoming plane wave. The distance between the bubble centers and the boundary is 20 times the bubble radius.

Fig 4a shows the comparison of the noise reduction for the cases of a single bubble and two bubbles. The amount of noise reduction is similar but the maximum value has moved
to a lower frequency. Moreover, the simulations show two main eigenfrequencies for the bubble cluster. One corresponds to the joint eigenfrequency, so that both bubbles oscillate in phase, as can be seen in Fig. 5. The second eigenfrequency describes the out-of-phase oscillation of the two bubbles. Fig. 4b shows the influence of the distance between the bubble centers on the value of the omnidirectional first eigenfrequency of the 2 degree of freedom (DOF)-system. It can be observed that the joint eigenfrequency decreases with decreasing distance. For widely separated bubbles, the first eigenfrequency coincides with that of a single bubble. At a distance of four radii the value of the eigenfrequency is reduced to 90% of that of a single bubble.

Fig. 4: a) Absolute pressure averaged over the right side of the control volume. b) Change of the omnidirectional first eigenfrequency of the 2 DOF-system over the distance

Fig. 5: Absolute pressure distribution in the control volume at resonance
4. CONCLUSION
In this paper a method based on the wave equation for the detailed modeling of a bubble curtain is presented. Two simple cases have been analysed: one with equidistant bubbles of the same size and one with bubble clusters consisting of two identical bubbles. It has been shown that the wave equation captures the effect of bubble oscillation under an incoming plane pressure wave. The noise mitigation effect was observed around the bubble eigenfrequency. Due to interaction between bubbles a shift of the eigenfrequency towards lower frequencies could be observed.

In next steps different bubble configurations and a higher air fraction will be considered. Preliminary results for bubbles with different sizes show different results with respect to the interaction between bubbles of the same size. In following steps realistic bubble distributions should be considered. To take the thermal and viscous effects into account the harmonic Navier-Stokes equations will be used as governing equations.

5. ACKNOWLEDGEMENTS
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REFERENCES

THE ROLE OF AIR BUBBLES IN ACOUSTIC SURFACE LOSS CONFIRMED BY HISTORICAL DATA (1949-2005) ABOUT ATTENUATION EXCESS OF SOUND IN OCEANIC SURFACE CHANNELS

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Abstract: Surface sound channels, associated with mixing layers, are frequently observed in ocean environments. Acoustic rays may remain confined by a barocline sound-speed gradient in the immediate neighbourhood of the surface and are impacted cumulatively by superficial phenomena that would remain negligible in configurations involving rare encounters with the sea surface. The most visible effect is a strong attenuation excess when compared with attenuation due to only viscosity and chemical relaxation in bubble-free seawater. Our presentation gives results from two topics related with this question: 1/ a synthetic review of publicly available experimental data concerning attenuation excess in surface channels; 2/ an attempt to get some global physical understanding and a numerical prediction of this excess, relying on models for specific features of the surface neighbourhood (surface waves and swells, stable populations of micro-air-bubbles).

Firstly, we summarise all published results. This amount of work, turned into a common representation, gives a very consistent, convincing picture of sound attenuation in surface channels, as a function of frequency and sea state. Secondly, we try to identify the physical phenomena mostly responsible for the observed attenuation excess. For this purpose, we compared the predictions of numerical models, including scattering from the rough moving sea surface and the effects of micro-air bubbles (Hall model for bubble population). Micro-bubble effects are sufficient for explaining the observed trends in global attenuation; we also consider the lesser but slightly noticeable impacts of probably surface roughness and large air bubbles in plumes associated with whitecaps.

Keywords: propagation, sea surface, surface loss, attenuation, air bubbles.
1. MULTIPLE SURFACE INTERACTIONS IN CONFINED SEA CHANNELS

Sea surface channels, associated with mixing layers, are primordial in the understanding the interactions of sound with the sea surface and its immediate neighborhood. The acoustic rays, which remain confined near the surface by upward refraction, are impacted cumulatively during their way from source to target; this cumulative effect enhances phenomena, that would remain unnoticed for deeper ray paths undergoing only one, or a small number of encounters with the upper boundary.

2. ATTENUATION EXCESS IN SURFACE CHANNELS

We firstly list all published results about experimental models of attenuation in surface channels, most notably three large US campaigns of the 1945-1975 era:

- [1]: AMOS data collected and analysed by Marsh and Schulkin in 1949-53;
- [2]: complementary NRL measurement analysed by Saxton-Baker (1950-55);
- [3]: later campaigns by Hall for validating previous data (Hall, 1972-76).

These data investigate the frequency interval 2 kHz-20 kHz. Later, smaller British and US British campaigns of the 60’s and 70’s provide further information, particularly for lower frequencies (down to a few 10 Hz):

- [4]: Kibblewhite & Denham, 1965;

All these authors observe an attenuation excess: in surface channels, as function of horizontal range, the acoustic intensity decays significantly more quickly than what can be predicted using the “classical” attenuation rate due to visco-chemical absorption only. Most of them propose synthetic formulas for describing their observations.

2.1 AMOS formula (Marsh & Schulkin):

Attenuation excess $\beta$ is given in loss per bounce, over the frequency range 2 kHz-25 kHz.

\[
\begin{align*}
\text{Regime } " \text{low } fkHzZ_{ft} " & \quad \text{Regime } " \text{high } fkHzZ_{ft} " \\
\text{if } fZ < T & \quad \text{if } fZ > T \\
\beta = 10. \log_{10} [ 1 + ( fkHzZ_{ft} / C)^4 ] & \quad \beta = D ( fkHzZ_{ft} )^{1/2}
\end{align*}
\]

Notations: $fkHz$ = frequency, $Z_{ft}$ = average trough-crest wave height, in feet, $\beta$ = loss excess per bounce, in dB

\[
C = \begin{cases} 
1.0 \text{ kHz m} & \text{3. dBkHz}^{-1/2}m^{-1/2} \\
3.35 \text{ kHz ft} & 1.64 \text{ dBkHz}^{-1/2}ft^{-1/2} \\
4.14 \text{ kHz ft} & 1.59 \text{ dBkHz}^{-1/2}ft^{-1/2}
\end{cases}
\]

\[
D = \begin{cases} 
1. \text{ kHz m} & (I.Schulkin version) \\
3.35 \text{ kHz ft} & (Schulkin version,1969) \\
4.2691 \text{ kHz ft} & (Weinberg version,1985)
\end{cases}
\]

Attenuation excess $\beta$ per bounce, when divided by the horizontal period $\Delta x$ of the critical ray, is turned into loss excess $B$ per range unit:

\[
\gamma \sim 1.6 \times 10^2 \text{ m s }^{-1} \text{ m}
\]

\[
\Delta x \approx \left( \frac{8 \frac{cH}{\gamma}}{Z} \right)^{1/2} Z^{1/2}
\]
2.1 Formula of Saxton & Baker: investigated frequency band 2kHz-7.5kHz

\[ \alpha = A + B \]

(unit: dB/kyd)

\[ \begin{align*}
\alpha &= \frac{1}{32.768 + f_{kHz}^{3/2}} \left( 1.776 f_{kHz}^{3/2} + f_{kHz}^3 \left( \frac{0.65053 f_{kHz}^2}{f_T^2} + \frac{0.026847 f_{kHz}^2}{f_T^2} \right) \right) \\
B &= \frac{26.6 f_{kHz}^{1.4}}{\left[ (1452 + 3.5 T_{ft}) Z_n \right]^{1/2}}
\end{align*} \]

Turned into a common representation in terms of attenuation per range unit, these two formulas give a very consistent, convincing picture of sound attenuation in surface channels, as a function of frequency, channel depth and sea state; Fig.1 displays this attenuation, as observed by Marsh and Schulkin (AMOS), and by Saxton and Baker (SB), as functions of frequency, for sea state 3, and for different channel depths (50m, 100m, 200m). Considering the uncertainty intervals, AMOS and SB are mutually coherent. For comparison, we also plotted on the same graphs the attenuation in bubble-free clean sea water predicted by the formula of François and Garrisson, emphasizing this way the excess of loss in surface channels. Similar conclusions may be drawn for sea states 1 and 2, and for shallower channels with depths down to 25m.

3. PHYSICAL MODELLING OF ATTENUATION IN SURFACE CHANNELS

We tried to identify the physical mechanism mostly responsible for the observed attenuation excess. For this purpose, we compared the predictions of numerical models, including scattering from the rough moving sea surface and the effects of micro-air bubbles; we adopted Hall’s model ([6]) for stable populations of air micro-bubble layers (bubbles with radii around 50µm, resonant around 70kHz). The presence of these micro-bubbles induces two phenomena over the first metres below the sea surface: first, a sharp decrease of sound-speed (bubbly water is more compressible than bubble-free water); second, a huge increase of attenuation. Even if limited to a very narrow layer of a few m’s only, these two mechanisms, when cumulated, have an important effect (deviation of rays near the surface compared with bubble-free water + attenuation excess). We developed as accurate as possible a propagation model dealing with these mechanisms and evaluated the decay of acoustic flux in the surface channels for comparison with the published experimental models introduced in the previous section.

Fig.2 displays the modeled attenuation as function of frequency, for sea state 3, and for different channel depths (50m, 100m, 200m), plotted over the empirical formulas fitted on experimental data by Marsh and Schuklin (AMOS) and by Saxton and Baker. The essential orders of magnitude and trends are remarkably reproduced. Beyond 7-to-8kHz, the agreement is excellent, whereas slight bias of attenuation toward lesser values may be observed at lower frequencies (typically, up to 8kHz); this means that micro air-bubbles, even if an essential and dominant cause of attenuation excess near the surface, are not alone in the global attenuation. There is little, but noticeable room for other low-frequency mechanisms.

We tried our modeling in another kind of channel with confinement near the sea surface: shallow waters, with little bottom attenuation (sandy or gravelly sea beds). Fig.3 displays the results of a comparison of loss predicted by our bubble model with at-sea observations by Weston (Weston and Ching, [8]). The same agreement than for surface channels may be observed, with the same slight bias toward lower values.
Fig. 1: Attenuation in surface channels observed by Marsh and Schulkin (AMOS) and by Saxton and Baker, as function of frequency, for sea state 3, and for 3 different channel depths (50m, 100m, 200m).
Fig. 2: Attenuation in surface channels observed by Marsh and Schulkin (AMOS) and by Saxton and Baker, compared with our modelling relying on air micro-bubble attenuation (Hall model).
4. CONCLUSION

Micro-bubble effects are sufficient for explaining most of the observed trends in global attenuation excess in surface channels and in shallow waters with hard bottom. Nevertheless, this mechanism seems to predict losses slightly lower than observations, particularly below some 7kHz. We have to consider the lesser but slightly noticeable impacts of other attenuating phenomena, a list of which may include the following items:

- scattering from surface roughness and spreading by surface movements;
- large air bubbles (with radii up to the order of 1mm and resonant at low frequencies of the order of a few kHz), present in plumes associated with whitecaps;
- populations of fishes with swimbladders.

REFERENCES

Session 7

Acoustics of marine renewable energy developments

Organizers: Stephen Robinson, Paul Lepper and Philippe Blondel
DESIGNING PRACTICAL ON-SITE CALIBRATION PROTOCOLS FOR ACOUSTIC SYSTEMS: KEY ELEMENTS AND PITFALLS

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Abstract: Although acoustic systems are increasingly being used for environmental and noise surveys of marine energy devices, there are currently no standard protocols for the on-site full bandwidth calibration of these systems. Reports often include little or no information on the methods of calibration used before, during or after surveys. Without proper calibration, the sound levels may be far from accurate, leading to skewed reporting and inaccurate conclusions.

Hydrophone calibrations from internationally recognised standardisation centres, such as NPL, allow providers to reference their systems to international standards. Marine renewable energy devices, however, are often deployed in remote areas and it is not always practical or cost-effective to send every acoustic system to be independently tested before every deployment. On-site referencing of multiple units to a single standardised system can help improve calibration traceability. Although this may at first appear relatively simple, the production of an accurate, full-spectrum calibration, particularly in real-world test sites, is surprisingly difficult.

Keywords: Calibration, Marine Energy, Standards, Reporting
1. INTRODUCTION

Calibration of acoustic systems can seem complex and is particularly difficult to achieve in remote areas, away from services for standard referencing of equipment. This is true for acoustic surveys of marine energy devices, which are often deployed on remote areas of coast. Setting up a portable calibration system can avoid these difficulties.

This paper will describe the key elements involved and pitfalls to avoid when setting up such a system. It will use work at the European Marine Energy Centre as an example, with particular reference to the whole system comparison calibration of Drifting Acoustic Recorder and Tracker (DART) units.

2. KEY ELEMENTS

The main focus for ensuring accurate calibrations should be to assess and reduce levels of uncertainty throughout the calibration system. Standards organisations have created guidance for both calibration techniques and the assessment of uncertainty. These provide extensive information to ensure accurate characterisation of acoustic systems in laboratory conditions, including good practice and pitfalls to avoid. It is recommended that these are consulted during detailed design of calibrations [1, 2, 3, 4].

For simplicity, these descriptions of uncertainty are typically grouped into type A and type B uncertainties. Type A are ‘random’ uncertainties; those which can be calculated through repeated measurement. Type B are ‘systematic’ uncertainties, which can be predicted and reduced through thoughtful calibration design.

This section will provide a brief overview of important considerations for initial calibration design. This will use as an example the comparison calibration design for the DART units, shown in Figure 1.

![Figure 1: Overview of DARTs comparison calibration, including known data (Orange) from referenced data (Pa/V) and acquired data (units), and unknown physical quantities and conversion factors (Blue) that require further measurement.](image)

Elements of the system (I-VII) are described in Table 1.
Whole system comparison calibration

Comparison calibration is relatively simple and fast, but requires a large body of water and a stable, calibrated reference hydrophone (see section 3 for details). These requirements can usually be achieved in remote marine areas, through access to the sea and annual reference calibrations. More information on alternative calibration techniques can be found in ISO standards and guidance literature [2].

The technique uses a source, emitting hydrophone and two receiving systems. The first receiver, the reference system, consists of a hydrophone and subsequent signal conditioning pre-calibrated to international standards to provide pressure in Pascals (Pa) from measured voltages, or \( \text{Pa/V} \).

The reference hydrophone is then placed next to a hydrophone from a second un-calibrated system, for which only Unit values (such as bits in a .wav file) are known. If both hydrophones are located at the same distance from the source, then the acoustic pressure (in Pa) at both hydrophones should be the same. It is therefore possible to use the Pa excitement of the reference system to calculate Pa/Unit values for the un-calibrated system.

However, there are a number of unknowns; for example, how is the voltage from the reference hydrophone converted to units? Typically this will involve the analog-to-digital converter (ADC) within a data acquisition unit. Thus the conversion factor of the ADC, the \( \text{Units/V} \) produced, will need to be estimated before Pa values can be calculated from measurements by the reference system.

The conversion factors and physical quantities highlighted in blue in Figure 1 are all unknown; by using additional measurements (see section 3), it is possible to calculate conversion factors, which will then provide a basis for calculating the physical quantities.

Estimating uncertainty

Uncertainty in measurement of any part of the process, including the measurement of conversion factors, could affect the accuracy of the whole calibration. Therefore, it is important to predict and test those elements that could create uncertainty. Table 1 outlines the expected uncertainties for each aspect of the DART's calibration, as further discussed in section 3.

As previously described, type A uncertainties are reduced through repeated measurements, whilst type B are typically reduced by modifying the system design. Often systematic uncertainties will not become obvious until the calibration is trialled, so it is useful to plan test calibrations well in advance of data gathering, in order to allow time for purchasing and testing any additional equipment if required.

When estimating uncertainty, it is important to convert all estimates and measurements to the same units. This can be simplest to achieve using percentages of uncertainty values. It is also useful to note that logarithmic units may make certain calculations more difficult; thus it may be helpful to work with linear units.
Table 1: Example of a simple uncertainty analysis, taking into account expected sources of uncertainty. Detailed includes actions that were successfully used to reduce uncertainty (Orange) and actions needed to further reduce uncertainty (Blue).

<table>
<thead>
<tr>
<th>Uncertainty Type</th>
<th>A (random)</th>
<th>B (Systematic)</th>
<th>Mitigation</th>
<th>Accuracy</th>
</tr>
</thead>
<tbody>
<tr>
<td>I. Signal generation</td>
<td>Waveform shape and levels may change</td>
<td>- Test signals and make waveforms as similar as possible</td>
<td>B -Signals tested</td>
<td>Orange - Positive outcomes; some analysis tools are not as sensitive as expected</td>
</tr>
<tr>
<td>II. Signal transmission</td>
<td>Random changes due to water movement, debris, etc.</td>
<td>Reflections, mounting vibrations, bubbles, other problems with setup</td>
<td>A -Repeat B - Avoid reflections/rig</td>
<td>Orange - Positive outcomes for A, B</td>
</tr>
<tr>
<td>III. Electrical/acoustic noise</td>
<td>Random electrical/acoustic noise</td>
<td>Equipment noise, noisy test area</td>
<td>A - Repeat B - Reduce equipment noise to minimum/Avoid noisy areas</td>
<td>Orange - Positive outcomes for A, B</td>
</tr>
<tr>
<td>IV. Device under test receiving (Pa-unit)</td>
<td>Random changes in Acoustic response Pa-unit transduction will not be linear &amp; may change over time</td>
<td>B - Repeated calibration of the whole system</td>
<td>B - Repeated calibration</td>
<td>Orange - Positive outcomes for B</td>
</tr>
<tr>
<td>V. Reference equipment receiving (Pa-V)</td>
<td>Random changes in Acoustic response May not be calibrated accurately &amp; Pa-V transduction may change over time</td>
<td>B - Re-calibrate often and track change</td>
<td>A&amp;B - Reference hydrophone and conditioner calibrated. To be recalibrated again asap.</td>
<td>Orange - Positive outcomes for B</td>
</tr>
<tr>
<td>VI. ADC accuracy (V-Unit)</td>
<td>Random changes in electrical response V-unit conversion will not be linear &amp; may change over time</td>
<td>A - Repeated calibration</td>
<td>B - Calibration of DAQ</td>
<td>Orange - Positive outcomes for A, B</td>
</tr>
<tr>
<td>VII. Analysis accuracy</td>
<td>Different methods of analysis will create different results for the same analysis (e.g. fft)</td>
<td>B - Compare and decide upon most accurate method</td>
<td>Orange - Positive outcomes for B</td>
<td>Orange - Positive outcomes for B</td>
</tr>
</tbody>
</table>
3. POTENTIAL PITFALLS

Signal generation and acquisition

The first potential pitfall for a calibration is the calibration tone; both the generation of the electrical signal and the transduction into a physical signal can produce frequency and amplitude artefacts. Therefore, it is important to check the calibration signal and modify the calibration system as necessary. For example, in Table 1, a system with better defined waveforms was required; therefore equipment with improved specifications was purchased. Ideally, the source hydrophone should also be calibrated across the frequency range of interest in order to reduce uncertainties in source amplitude. Generation and acquisition systems also need to be designed to cover all frequencies required for acoustic surveys. For example, if surveys require measurement up to a frequency of 200 kHz then it will be necessary to use a generation and acquisition system capable of working at these frequencies, i.e. an acquisition system with a sample rate of at least 400 kHz.

Signal transmission

A major influence upon the transmission of the signal is the environment. Uncertainty due to random changes can be reduced using multiple repeats at each frequency. However, systemic effects, such as the effect of boundaries in the test area, must be designed into the calibration. For example, hydrophones must be placed at a depth such that reflections from the surface and bottom reach the receiving hydrophones a certain period after the direct signal. In Table 1 (VII), the calibration pulse is analysed using a 512 Fourier transform, thus a period of at least 512 samples will be needed before the first reflection is received. The time between direct and reflected signals is related to the distance between the source and receiving hydrophones, since a shorter distance will provide more time before reflections are received. However, this will increase the minimum frequency available for testing. This frequency can be calculated, based upon estimates of the 'far-field' [2], using:

$$\text{MinFreq} = 2 \times \left( \frac{c}{\text{distance}} \right)$$

(10)

with distance in meters and c as speed of sound (in m/s) within the local environment. It is therefore important to realise that the typical minimum frequency for a comparison calibration will be low kHz and other methods must be used for calibrating low frequencies, such as a pistonphone calibration [2]. It may be necessary to reduce hydrophone distance, and so increase the minimum frequency, in shallow test areas.

Speed of sound itself can be calculated using CTD measurements of the calibration site. Guidance on the measurement and calculation of speed of sound can be found on the NPL website [5].

Finally, the positioning, mounting and wetting of the hydrophones can influence transmission. Inaccurate positioning can change the amplitude of signals, whilst mounts
can create vibrational artefacts. Therefore it is important to state and check the hydrophone distances and use guidance to avoid mount vibrations and air bubbles [6, 7].

**Additional measurements**

Additional measurements, such as CTD measurements, will also require calculation of uncertainty, typically through calibration techniques or using standardised calibrated equipment. The importance of these calibrations will depend upon the probable effect on calibration; for example, inaccurate reference hydrophone calibration or data acquisition equipment response calibration could greatly affect values, whereas inaccurate CTD measurements may not have a great effect.

**Equipment accuracy**

The accuracy of acquisition equipment and the equipment used to test this equipment must be checked over time and if calibration methods change. For example, data acquisition equipment can be tested with voltage signals calibrated using multi-meter measurements. Systematic uncertainties of electrical equipment can be avoided through regular testing and comparison of voltage and unit values; for example, hydrophone connections may produce different values if connected before or after powering elements of the system.

**Calculation of values**

It is useful to check all calculations performed on measurements manually before any automation is introduced into the calibration process. There are a number of steps required for calibrations and it is essential to ensure that each step is calculated correctly. This is particularly important when using analysis packages to calculate the amplitude of received pulses since the values can change depending upon the section of pulse analysed and the length of section analysed. Ideally, the section analysed should be as long as possible, although this may not be feasible in practice. Finally, it is good practice to perform these calculations in SI units before converting to dB; as previously mentioned use of logarithmic units can complicate certain calculations. It is also good practice to use RMS amplitudes and clarify this use in reports; this avoids comparison of peak and RMS amplitude values, which, although proportional, are not equivalent.

4. **CALCULATING UNCERTAINTY**

The method for calculating uncertainties differs for each type of uncertainty. In addition, the following methods assume that elements are independent of each other, whereas correlated elements require additional calculation [3].

**Calculating type A uncertainties**

These uncertainties are estimated through the calculation of arithmetic mean and standard deviation of repeated measurements. Guidelines recommend between 4-10 repetitions to cover most measurements, although accuracy can be slightly improved with additional repeats [1].
Arithmetic mean ($\bar{\chi}$) is simply calculated using:

$$\bar{\chi} = \frac{\sum \chi_i}{n}$$  \hspace{1cm} (1)

where $n$ is the number of repeats, and $\sum \chi_i$ is the sum of their values.

Standard deviation ($\sigma$) is then calculated using:

$$\sigma = \frac{\sum (\chi_i - \bar{\chi})}{n - 1}$$  \hspace{1cm} (2)

Percentage standard deviation can then be calculated:

$$\sigma(\%) = \frac{\sigma}{\bar{\chi}}$$  \hspace{1cm} (3)

Finally standard type A uncertainty ($u_a$) is calculated using:

$$u_a = \frac{\sigma(\%)}{\sqrt{n}}$$  \hspace{1cm} (4)

**Calculating type B uncertainties**

Type B uncertainties are estimated using the best information available and experience of the systems being calibrated. In practice, an estimate is made of the upper and lower bounds of expected error, the difference between these values taken and divided by 2:

$$a = \frac{(\text{upper} - \text{lower})}{2}$$  \hspace{1cm} (5)

The mean is typically half-way between these two values:

$$\bar{\chi} = \text{lower} + a$$  \hspace{1cm} (6)

Standard uncertainty can then be calculated based upon the expected distribution of the error. For example, if all values across the range from the upper to lower estimate are expected to be equally probable, then a rectangular distribution can be used:

$$u_a = \frac{a}{\sqrt{3}}$$  \hspace{1cm} (7)

whilst errors expected to cluster around mean, i.e. in a bell-shape, can be calculated using:
\[ u_p = 1.48a \] (8)

Finally, the percentage uncertainty can be calculated as above.

**Calculating combined uncertainties**

Once the uncertainty for each element has been calculated, it is possible to calculate the combined uncertainties, \( u_c \), through the formula:

\[ u_c = \sqrt{u_1^2 + u_2^2 + u_3^2 + ...} \] (9)

This formula is useable for uncertainties that combine linearly to produce a total uncertainty. This is not generally the case for calibrations, since each uncertainty is typically independent of others, but it can provide a simple 'worse-case scenario' estimate.

Additional, more complicated, means to calculate combined uncertainty are detailed in international standards for uncertainty and measurement [3].

**5. ACKNOWLEDGEMENTS**

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**REFERENCES**


AN ENVIRONMENTAL SURVEY AROUND THE NAREC OFFSHORE ANEMOMETRY HUB (NOAH) – A COMPARISON BETWEEN ACOUSTIC MEASUREMENT INSTRUMENTS.

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\textbf{Abstract:} In July 2013 a survey was conducted covering the area around the Narec Offshore Anemometry Hub (NOAH), 3km offshore of Blyth in North East England. The site will eventually host an array of 12 experimental wind turbines and the object of this baseline survey was to provide an initial dataset for analysis and also enhance understanding of instrument operation and sensitivity. The survey was carried out by members of the Bio-Acoustic Research Consortium (BARC) using the Newcastle University research vessel RV Princess Royal. A number of acoustic (pressure) sensors/recorders were deployed, some of which were built in-house by consortium members and some of which were Commercial Off The Shelf (COTS) devices, along with a particle velocity sensor as well as some grabs and trawls for collecting various species for lab-based behavioural analysis. A visual survey was also undertaken along a zig-zag transect for comparative analysis of detection rates. This paper will describe a comparison between the acoustic measurements obtained with two COTS instruments: a Songmeter SM2M (Wildlife Acoustics) and SSQ906G LOFAR sonobuoys (Ultra Electronics), along with results obtained with the particle velocity sensor. The analysis covers a total duration of 2.5 hours of simultaneous recordings from the SM2M and sonobuoys. The paper will present examples of sounds recorded with both instruments, along with power spectra and other analyses and will explore the reasons for the differences.

\textbf{Keywords:} Passive Acoustic Monitoring, Noise, Instrumentation
1. INTRODUCTION

The In 2010, the UK government launched its 3rd round of offshore wind farm deployments around the British Isles. On this occasion, the National Renewable Energy Centre (Narec) began setting up a demonstration facility for pre-commercial testing in real conditions named the Offshore Wind Demonstrator Project. This test site (situated close to Blyth, Northumberland) will allow industries to measure and optimise the design of turbines of up 7MW. The first stage of the project was deployment of the Narec Offshore Anemometry Hub (NOAH) to validate environmental and wind resource measurements. This was installed in 2012, three nautical miles offshore from their facility in Blyth.

In 2011, a group of engineers and scientists from academia and industry formed the Bio-Acoustic Research Consortium (BARC) to undertake underwater acoustic monitoring of marine mammals, ambient and anthropogenic noise prior to, during and after the installation of the anemometry hub NOAH and the subsequent installation of the turbines. This consortium aims to reduce uncertainties around underwater noise monitoring and impacts at marine renewable energy sites. The group also focuses on comparison of different instrumentation and methodologies to measure underwater ambient and anthropogenic noise. This will enable the regulators to agree on common standards and provide guidance regarding underwater acoustic monitoring at these sites.

2. THE 2013 SURVEY

The underwater ambient noise monitoring considered here was undertaken during the wind farm pre-construction phase to establish a baseline for the underwater noise levels and biological and anthropogenic activity in the area during the summer of 2013. A number of instruments, including the SM2Ms discussed in §Error! Reference source not found., were deployed semi-permanently over the period whilst others, including the sonobuoys described in §Error! Reference source not found. and the accelerometers in §Error! Reference source not found., were deployed for short periods from the Newcastle University research vessel Princess Royal. Fig. 1 shows the Princess Royal
track and the locations of various instrument deployments, as described in more detail below.

3. INSTRUMENTATION AND ANALYSIS

3.1. SM2M

An autonomous recording acoustic device, the SM2M+ Ultrasonic from Wildlife Acoustics was deployed at approximately 100m from the NOAH hub, at 55°08.836N, 1°25.334W. It was located 1.5m from the sea bottom (water depth ~44m).

The recordings were obtained using a 384 kHz sampling rate to capture high frequency biological sounds such as clicks from harbour porpoises (Phocoena phocoena). Data were collected during 17 days, recording continuously using an HTI-99 ultrasonic hydrophone, sensitivity -165dB re: 1V/µPa and a gain setting of 12dB. The device was programmed using Song Meter SM2 Firmware to record in 30-minute intervals, allowing for easier analysis by reducing file sizes.

Some of the data were compressed to WAC0 format to maximise use of the hardware space and deployment time. Any WAC0 compressed files were converted into WAV format using Kaleidoscope software for further analysis. Each full 64GB of data in WAC0 format took approximately 10 hours to convert to WAV format. The laptop used for the analysis had an Intel processor i5-2520M 2.50 GHz, dual core.

Adobe Audition was used to investigate and classify the different types of sounds (anthropogenic noise and biological sounds) found in the recordings, in accordance with the three indicators used to determine GES when monitoring marine noise. Sounds were classified as low/medium frequency impulsive (indicator 1), high frequency impulsive (indicator 2), continuous (indicator 3), and as biological. This report focuses mainly on indicators 1 and 3. The contribution of each of these categories to the overall noise distribution was quantified. This screening of sounds took approximately 8-10h for each 24h of recording. The time taken for analysis varied depending on the level of activity in the sample. Analysis could take over 2 hours for a 30-minute section if a lot of biological activity was recorded.

To understand the contribution of the various sources in terms of sound levels, and how these change daily, the Sound Exposure Level (SEL) was calculated over different periods of time (5, 10 and 15 seconds) using the following equation:

$$SEL = 10 \log_{10} \left( \frac{E}{E_0} \right) \text{ dB re } 1\mu Pa^2 \text{ s}$$

Where the sound exposure $E$ is:

$$E = \int_0^T p(t)^2 \, dt \quad \text{and} \quad E_0 = p^2 T_0$$

$p_0$ is the reference sound pressure in µPa, $T_0$ the reference time in s and $T$ the averaging time.

The interpretation of these calculations would form the basis of an investigation into the potential for an appropriate duty cycle to be established. This could reduce the amount of data collected each day without losing relevant information regarding the sound level.
changes over an extended period of time. However, this type of analysis requires a significant computational effort when dealing with such high frequency sampling rates and, therefore, it was not possible to pursue the matter beyond the 15 sec calculation with the software available. The calculation of the SEL for 5, 10 and 15 second periods took around 50 hours to complete for the full dataset.

Whilst the full dataset was used to assess the soundscape of the area, the data available that overlapped with the other recording devices was used to compare the capabilities of the systems. The SM2M ultrasonic is designed as a platform to monitor biological signals rather than underwater noise and its characteristics are not optimised to the task of monitoring underwater noise.

### 3.2. Sonobuoys

The SSQ 906G LOFAR Sonobuoy has been developed for military purposes to detect submarines. It is compact (length: 41.9 cm, diameter: 12.3 cm) and lightweight (5.6 kg) and thus easy to transport and deploy from an aircraft or a vessel. As it has been designed to operate for one to seven hours, so no long-term data can be collected in contrast to SM2M recording devices. The sonobuoys used in this study consisted of a single omnidirectional hydrophone (frequency range: 10 – 20,000 Hz) on a coiled, low self-noise cable, a radio transmitter and an antenna attached to a surface floater to allow real-time data transmission and monitoring. The system is designed to drift freely with the water current to reduce hydrodynamic low frequency sounds. A pre-whitening filter further reduces low frequency sound levels and enhances dynamic range.

Ambient sound recordings discussed here were made on July 2 from 12:44 - 13:05 at 55°11.514’N / 1°27.565’W (sea state: 4, water depth: 40, no rain) at the same deployment site as the accelerometer (§Error! Reference source not found.), 3 nm away from the deployment site of the SM2M recorder (§Error! Reference source not found.), and July 3, 2013 from 13:49 – 16:33 at 55°11.322’N / 1°27.799’W (sea state: 0, water depth: 36, no rain). Recordings were made using SSQ 906G LOFAR Sonobuoys (Ultra Electronics, Greenford, UK) deployed from Princess Royal using a sampling rate of 48 kHz and 32 bits. Radio-transmitted acoustic signals were received using WiNRADiO G 305 v2.22 (20013 WiNRADiO Communications, RADIXON UK Ltd, Chesterfield, UK) and recorded on a laptop computer (Toshiba Tecra M11-17V, Toshiba Information Systems UK Limited, Weybridge, UK) using Audacity v2.0.3 (1999-2013 Audacity Team; SourceForge.net).

Original noise levels were restored using an of the pre-whitening inverse filter in Adobe Audition v1.5 (1992-2004 Adobe Systems Incorporated) before analysis.

### 3.3. Accelerometer

Fifteen minutes of particle acceleration data were recorded at each of six different locations over two days with a tri-axial accelerometer (M30, sensitivity 0–3 kHz, manufactured and calibrated by GeoSpectrum Technologies, Dartmouth, Canada; recorded on a laptop via a USB soundcard, MAYA44, ESI Audiotechnik GmbH, Leonberg, Germany). Water depths ranged from 38-53 m and recording depth was always 20 m. Sea state was 2-4, wind speed 12.3-22.1 kt, water temperature 11.65-12.24°C at 8 m depth and salinity 31.8-32.3 ppt. The accelerometer was attached to a surface buoy that drifted with the Princess Royal.

Acoustic recordings were analysed in MATLAB v2010a. Recordings were split into 1 s
windows that were Hamming windowed. Fast-Fourier Transform (FFT) was performed on each 1 s subsample to translate the data into the frequency domain. FFT length was set so that an absolute value for every 1 Hz could be obtained for each second of recording between 0 and 22.05 Hz (the Nyquist frequency). These values were squared to obtain the power spectral density, multiplied by two and divided by 1.36 to correct for the noise power bandwidth of the Hamming window. The results were converted to dB re 1 (µm/s²)/Hz and used to plot spectrograms. The raw values were also used to calculate mean, median, 5th and 95th exceedance levels (equivalent to percentiles). Sound levels are only considered up to 3 kHz due to the upper limit of the sensitivity of the accelerometer. This frequency range is likely to cover the hearing range of most fish. Only the two horizontal axes are considered here as wave action meant that the vertical axis was too noisy. On one other occasion a loose wire meant that one of the horizontal axes was too noisy to analyse.

4. RESULTS

4.1. SM2M

The activity recorded at NAREC hub on the 3rd of July 2013 showed a variety of sound sources consisting mainly of boat activity during the day and biological sources, including harbour porpoises, dominating during the evening. Most biological sounds were recorded at very high frequencies, i.e. above 100 kHz, whereas anthropogenic sources dominated the lower frequencies.

![Fig. 2: Events recorded by the SM2M on July 3rd.](image)

The average duration of each event recorded during the 3rd of July is presented in the Table below. On average biological sound sequences lasted less than 2 minutes whilst anthropogenic sounds lasted much longer.

As only 2 and a half hours of data overlapped with recordings made using other systems, this time window was examined in more detail and the sound exposure level calculated with different time averaging period to investigate whether changing such parameter had an effect on the measured levels.
<table>
<thead>
<tr>
<th>Row Labels</th>
<th>Count of Type of sound</th>
<th>Average of Duration of event_MIN</th>
<th>StdDev of Duration_MIN</th>
</tr>
</thead>
<tbody>
<tr>
<td>03/07/2013</td>
<td>Biological 80</td>
<td>1.290069794</td>
<td>4.270973927</td>
</tr>
<tr>
<td></td>
<td>Continuous 40</td>
<td>17.79032211</td>
<td>11.45907516</td>
</tr>
<tr>
<td></td>
<td>Impulsive 32</td>
<td>10.25270595</td>
<td>12.17497658</td>
</tr>
</tbody>
</table>

*Table 1: Duration of events recorded by the SM2M on July 3rd*

**Fig. 3: SELs computed for averaging times of 5, 10 and 15 sec for SM2M recordings.**

A good fit in noise levels is observable for different averaging times; however, during the analysis times, no particular noises were recorded and also the difference in averaging times is very limited as it was constrained by the computational capacity of the system.

### 4.2. Sonobuoys

Since sonobuoys use radio transmission for real-time data transfer designed for human hearing range, sounds of up to 15,000 Hz could be recorded. There were no qualitative differences in sound recordings between the two days, but signals sent by the sonobuoys were lost sooner on the first day at sea state 4 (within minutes after the research vessel moved away) because of shorter transmission ranges than on the second day at sea state 0. Highest spectral levels of ambient sound ranged from 180 to 2000 Hz in the absence of vessel noise during recordings when the research vessel was drifting with engines turned off. These recordings also contained sounds associated with the research vessel interacting with waves. These sounds consisted of two main types:

1) short (up to 0.16 s) broadband sounds of sudden onsets with highest spectral levels ranging from 100 to 2000 Hz, and
2) sounds of slightly longer duration (up to 0.50 s) and lower frequencies ranging from 100 to 500 Hz.

Most of the recordings, however, contained ship noise mainly originating from the research vessel Princess Royal and some from passing cargo ships. Main sound energy from sounds emitted when the motor of the research vessel was started again after drifting ranged from 10 to 1100 Hz. When moving away, highest spectral levels reached up to
3000 Hz but fell with increasing distance to the sonobuoy to ranges from 10 to 750 Hz. Main sound energies from sounds emitted by vessels passing the sonobuoys within 100 m ranged from 10 to 1000 Hz, but spectral levels were increased throughout the whole sound spectrum of up to 15,000 Hz compared to ambient spectral levels without ship sounds. In none of the recordings, marine mammal sounds were detected, but very short (0.02 - 0.04 s) ‘clacking’ sounds of undefined origin, which may have originated from collapsing air bubbles generated by the research vessel’s propellers. Frequency ranges of these sounds were highly variable and mostly found within 1500 to 6000 Hz, but some consisted of lower broadband sounds with ranges from 10 to 4500 Hz.

4.3. Accelerometers

Particle acceleration was recorded at 6 locations (see Fig. 1). The recordings at locations 1 and 3 captured other boats at 0.73 and 1.8 nm respectively. Presented below are results from location 1.

![Power spectral density (PSD) of the square root or the summed squared two perpendicular horizontal axes of particle acceleration from a recording taken at location 1 (see map in Fig. 1) on 1/7/13 at 13:03.](image)

*Fig. 4:* Power spectral density (PSD) of the square root or the summed squared two perpendicular horizontal axes of particle acceleration from a recording taken at location 1 (see map in Fig. 1) on 1/7/13 at 13:03.
5. DISCUSSION AND CONCLUSIONS

This paper has presented an overview of some of the environmental measurements carried out at the NOAH site and given a qualitative description the results obtained, especially relating to a comparison between the SM2M, LOFAR sonobuoys and accelerometer recordings. It is hoped to publish a more quantitative comparison shortly.

It is immediately apparent that an important difference is the recording bandwidth. A sampling rate of 360 kHz for SM2M recordings enable detection of marine mammal clicks. The sonobuoy’s audio bandwidth is limited by the FM radio communications link.

Another difference is the usage of the two systems and the resulting advantages and disadvantages, e.g.:

1) SM2M: long-term recordings from a fixed location, data storage within SM2M.  
   Advantage: long-term recordings of frequencies up to 180 kHz  
   Disadvantage: Data can only be viewed after retrieving the recorder, no real-time control of data recording. Thus, week-long data recordings could be lost if data recording and/or storage did not work properly or recorder could not be retrieved.

2) Sonobuoy: deployment from a research vessel for real-time data recording via radio transmission for up to 7 h, this enables real-time short-term acoustic monitoring  
   Advantage: direct control of recorded data and online quality monitoring.  
   Disadvantage: signal could be lost quickly in rough sea states as radio-transmission can be easily interrupted by high waves (when receiver is on a vessel, not an aircraft), so long-term data recording may not be possible

Since there was constant vessel noise in the recordings, except for the short period on July 2, we can not make comparisons between flow noise recorded by sonobuoys vs flow noise recorded by SM2M devices to reliably show that the free-drifting design of sonobuoys can minimise water current noises.

The recording device to use will depend on the kind of data to collect. If it is important to collect real-time data for a short time span to control/monitor noise emissions during a specific operation (or monitoring marine mammal occurrence emitting low frequency sounds), usage of sonobuoys will be ideal (if operated from a drifting boat to avoid masking ship noises from the research vessel itself). If long-term surveys of ambient

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Fig. 5: Spectrogram of one horizontal axes of particle acceleration from a recording taken at location 1 (see map in Fig. 1) on 1/7/13 at 13:03.
sound levels, human noise emission over a longer time-scale or surveys of marine mammal occurrence emitting ultrasonic sounds are needed, SM2M will be the recording devices to use.

6. ACKNOWLEDGEMENTS

We firstly acknowledge the National Environmental Research Council (NERC) for their financial support for the BARC consortium and to Annie Linley for making this interdisciplinary collaboration possible. Additionally, we thank Newcastle University for the use of the Princess Royal and also Narec for giving us the opportunity of carrying out the measurements alongside their project.
ENVIRONMENTAL INVERSION WITH AN AUTONOMOUS HYDROPHONE IN A WAVE ENERGY DEVICE DEPLOYMENT SITE

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Abstract: This paper presents environmental inversion results of acoustic data in shallow water in Peniche (Portugal), collected in September 2013, during the Simple Underwater Renewable Generation of Electricity (SURGE) Project - a FP7 European collaborative demonstration project aiming at building a grid connected wave energy converter of type WaveRoller. A single autonomous hydrophone was moored at the position foreseen for the WaveRoller deployment. Computer generated acoustic signals were transmitted over a 3 km oceanic transect with a step of 300 m. Incoherent transmission loss (TL) for 1/3 octave frequencies in the band from 318 to 1270 Hz is used in an acoustic inversion procedure for the estimation of geoacoustic parameters of a two-layer seafloor. The acoustic inversion is posed as an optimisation problem aiming at minimising the root mean square error (RMSE) between field TL and replica TL, using a genetic algorithm. This procedure is repeated a number of times in order obtain a posterior distributions for each unknown parameter, and the solution of the inversion is obtained by taking the maximum of the a posteriori distribution of each unknown parameter. The RMSE between the field TL and acoustic model TL obtained for the solution across the acoustic transect varies between 1.5 and 2.6 dB, depending on the frequency. As a procedure to validate the obtained model, the RMSE between field TL and model TL is calculated for an alternative frequency band (from 1600 to 8064 Hz), in order to check the amount of model mismatch for frequencies not entering the inversion procedure. In that case, the RMSE varies between 2.5 and 4.6 dB. This increment in the RMSE can be considered relatively small, allowing the obtained physical model to be considered meaningful, and therefore adequate for noise modelling purposes over an eventual impact area.

Keywords: acoustic inversion, acoustic modelling.
1. INTRODUCTION

It is well established that the implementation of large-scale wave energy farms may have significant contributions in terms of introduction of underwater noise into the marine environment. The implementation of offshore renewable energy farms includes, in general, an acoustic monitoring plan which can be based both on in situ noise measurements and noise modelling studies. Noise propagation modelling offers the possibility to predict the soundscape over hypothesized scenarios with a quasi continuous spatial coverage, as to generate 2D noise maps at multiple depths.

A number of existing acoustic propagation models can provide the acoustic response of given environmental scenarios, provided that an accurate physical description of the propagation channel is available. The underlying physical model used in computational acoustic propagation models is typically made of a water column and one or multiple seafloor layers, and therefore the input may consist of parameters such as water depth, sound-speed profiles in the water column, and seafloor parameters such as sound velocity, density, attenuation, and thickness of one or multiple seafloor layers. Often, a complete and compatible description for a physical model with such a configuration is not available.

One option for assessing missing environmental data is by means of the so-called acoustic inversion [1]. An acoustic inversion is a procedure where physical properties are determined from experimental field data, by matching some acoustic observable with model replica data. Once a physical model has been determined and the noise source is accurately characterised, the user becomes able to predict the propagation of noise radiated away from the source. In the actual scope, this procedure can be seen as a field calibration.

This paper reports on acoustic inversion results obtained within the Project Simple Underwater Renewable Generation of Electricity (SURGE) in September 2013 in Peniche, off the West Coast of Portugal. Acoustic tones within the frequency-band 318 to 1270 Hz, transmitted over a 3 km transect, were received at a single hydrophone moored at the shallower end of the transect in order to estimated the acoustic transmission loss (TL). The data processing consists on the inversion for environmental parameters using a technique similar to classical Matched-Field Processing (MFP), using incoherent TL. The acoustic data inversion is posed as an optimisation problem, where solution search is carried out by means of a genetic algorithm (GA), and the final solution is determined by means of a posteriori probability distributions [2] generated from multiple runs of the GA.

The inversion results indicate that TL as a function of range can provide sufficient discrimination for inferring geometric and seafloor parameters, suggesting that this scheme can aid in overcoming missing environmental information relevant for noise propagation.

2. EXPERIMENTAL CONFIGURATION

2.1. Experimental configuration

The experimental configuration used for field calibration consisted of a transect with northwest direction departing from P1, with transmission positions M02 through M10 (see Figure 1). The mooring at P1 contained an autonomous hydrophone, a digitalHyd SR-1 by MarSensing, and the acoustic source was a Lubell LL916C. Figure 2 depicts the experimental
setup, with an autonomous hydrophone moored at 8 m depth, and an acoustic source tethered from the vessel down to 12 m depth. The acoustic transmissions were started at M02 (about 450 m from P1) and finished at M10 (about 3.05 km from P1).

![Experimental setup: mooring with an autonomous hydrophone and setup for source operation tethered from boat.](image)

**Figure 1:** Experimental configuration followed for field calibration. Acoustic transmission stations are indicated with triangle, and hydrophone moorings are indicated by squares.

**Figure 2** Experimental setup: mooring with an autonomous hydrophone and setup for source operation tethered from boat.

### 2.2. Acoustic signals

Field calibration is based on the transmission of a computer generated sequence, usually consisting on continuous waves, such as pure sinusoids, and LFM. In the present case a mix of different signal types was transmitted, among them, 8 s sinusoids with 1/3 octave band centre frequencies from 126 to 10160 Hz. Figure 3 shows the tones’ amplitudes estimated from a recording taken 1 m away from the source.

![Figure 3: Amplitude estimates for pure sinusoidal tones recorded with a hydrophone 1 m away from the source. The numbers along the plot indicate the respective frequency in Hz.](image)
3. EXPERIMENTAL CONFIGURATION

3.1. The baseline model

One of the most stringent issues for a model based inversion procedure is the choice of the underlying physical model. Often not all relevant physical properties are available. In the present case, part of the bathymetric data was readily available. The water depth of the most distant transmission positions were measured during the sea trial. Sound speed profiles were obtained in situ from a CTD device owned by WavEC. Figure 4 shows the baseline model reflecting the knowledge on bathymetry, water temperature, and emitter/receiver geometry. This is a three-layer model consisting of a watercolumn, a sediment layer and an infinite half-space. The bathymetry of the transect is highly range-dependent in water depth, as it goes from 12 m at the source position down to 41 m at the last position of the transect. The sound-speed profile was measured at position M10, the deepest water depth of the transect, presenting a typical Summer profile for the Portuguese Coast.

![Figure 4: Baseline model for acoustic propagation modelling.](image)

The water temperature at the surface was approximately 17 degrees Celsius, and 13 degrees Celsius at the bottom. The seafloor layers are parameterised with compressional speeds, densities, and attenuations. The compressional speed in the sediment is linear with depth. In this study the seafloor densities are defined as a function of compressional speed according to the following dependency [3]:

\[
\rho(c) = 14.8c - 21014 \quad \text{if} \quad c \leq 1.53
\]

\[
\rho(c) = 1.135c - 0.190 \quad \text{if} \quad c > 1.53,
\]

where $c$ is a compressional speed in the sediment or sub-bottom layer in km/s. This allows for melding two free parameters into a single parameter of sound speed with increased influence.
to the acoustic propagation, and a dependent parameter $\rho(c)$. Additionally, that knowledge can contribute to increase the solution constraint and to eliminate non-physical solutions from the search space. The baseline model shown in Figure 4 also shows emitter/receiver geometry. Note that these appear swapped, since for acoustic model computations, it is more convenient to provide one acoustic source and multiple receivers.

### 3.2. The objective function

The acoustic transmissions were designed having in mind an acoustic characterisation of the environmental medium in terms of transmission loss (TL) experienced by a continuous wave of a single frequency when it travels across the medium between emitter and receiver. In this case incoherent TL is considered, i.e., only absolute amplitudes at emitter and receiver are taken into account. The estimator for the TL is given as

$$
\hat{TL}_{inc}(f_k, r_n) = \frac{|\hat{X}_{inc}(f_k, r_n)|}{|\hat{S}_{inc}(f_k)|}
$$

where $f_k$ is the $k^{th}$ tonal frequency, and $r_n$ is the source-receiver range at the $n^{th}$ transmission position over a given transect. $|\hat{X}(f_k, r_n)|$ is the estimated amplitude of the received sinusoid transmitted from position $n$ of the transect, and $|\hat{S}(f_k)|$ is the estimated amplitude of the received sinusoid, both at frequency $f_k$. For a given transect, the TL is estimated for each source/receiver range with $n = 1, \ldots, N$, over frequencies $f_k$, with $k = 1, \ldots, K$. The objective function used for acoustic data inversion is based on the TL observed over the set of transmission stations considered for the ocean transect and a set of frequencies. For this study the following error function was adopted:

$$
E(\theta) = \sqrt{\frac{1}{KN_p} \sum_{n=1}^{N} \sum_{k=1}^{K} [TL_{db}(f_k, r_p, \theta) - \hat{tl}_{db}(f_k, r_p)]^2}
$$

where $TL_{db}(f_k, r_p, \theta)$ represents the TL observed between the emitter and a receiver at range $r_p$ and frequency $f_k$, while $\hat{tl}_{db}(f_k, r_p, \theta)$ represents the replica TL for a candidate parameter vector $\theta$. This objective function is the root mean square error (RMSE) between the two quantities described above over distance and frequency. The idea is to match the observable both over space and frequency, as an attempt to exploit the diversity available over these domains, in order to cope with the solution ambiguity inherent to a large number of free parameters.

### 3.3. Data processing

The acoustic inversion is roughly divided into three steps: first, the transmission loss is estimated using the acoustic field received for each transmission station and the spectral amplitudes of the emitted signal. This step generates $\hat{tl}_{db}(f_k, r_p)$ in eq. (3).

The second step is the main step of the inversion procedure. The search parameters are divided into sediment layer parameters (upper and lower compressional speeds, wave attenuation, and sediment thickness), sub-bottom parameters (compressional speed, wave
attenuation), and geometric parameters (source and receiver depths). Note that source and receiver swap roles, where inversion will proceed as if the acoustic source was deployed at a fixed position and multiple receivers were placed across the transect. Candidate acoustic models will be computed with the KRAKEN normal-modes computer code [4]. Replica TL is matched with the observed TL using eq. (3). For each inversion 10 independent GA populations were started. The GA was set to 40 generations of 70 individuals. The mutation probability was set to 0.008, in order to cause 30% of the population to be mutated at each generation, provided that each individual is coded into a bit chain of 39 bits. The crossover probability was 0.8. The size of the search space is approximately $5.5 \times 10^{11}$. Since the GA is a stochastic search method, for each data set multiple independent searches are carried out.

The third step aims at obtaining a final estimate by merging candidate solutions of the last generation of each population into marginal empirical probability density functions (PDF) of each parameter. These empirical functions are obtained by summing the fit of the final candidates over each parameter search interval. The empirical distribution will depend on the distribution of the candidates and their fits. These empirical distributions can provide a statistical analysis on the solution convergence, as one can calculate the mean value and the variability over the search interval or maximise the distribution in order to obtain the model estimate. This model estimate has been called maximum a posteriori (MAP) solution (in a Bayesian framework) [2].

### 3.4. Inversion results

The inversion was carried out considering frequencies 318, 400, 504, 635, 800, 1008, 1270 Hz for matching replica TL with observed TL using eq. (3). The number of receiver positions is 9, i.e., positions M02 to M10. The search consisted in matching the TL predicted with the acoustic was against the TL estimated from the acoustic data collected in the real environment for each pair range/frequency. The search was performed by a global optimisation scheme which is stochastic, resulting in a different solution for each search. Therefore the final result is obtained as a statistical observation. Figure 5 shows the empirical marginal distributions for each parameter based on 12 runs of the optimisation procedure. The geometric parameters (left column) present compact a posteriori distributions, with a maximum for source depth at 10.4 m, and approximately 5 m for receiver depth.

![Empirical probability functions for eight parameters.](image)

In the middle column are shown the distributions for sediment. In general, these distributions are compact, presenting reduced ambiguity. The upper speed in sediment has the MAP at 1683 m/s. This is in line with typical sound speeds in sandy seafloors (1650 m/s). The wave
attenuation has MAP at 0.45 dB/λ, also in line with table values for sand. The distributions for sub-bottom show more spreading than those for sediment, which is expected due to the low sensitivity of the field to deeper layers. Nonetheless, there is single peak for sub-bottom speed, and some ambiguity for sub-bottom attenuation. The set of parameters obtained by means of the inversion procedure is now seen as a physical model that could be used in a noise modelling procedure within that area. In order to have an idea of the accuracy of the

model, replica TL generated by the computational model, over range, can be compared with in situ measured TL. Figure 6 shows the TL data for frequencies 318, 504, 800, and 1270 Hz in comparison with the replica TL computed for the best model of each independent population, and the replica TL computed for the MAP estimate. Also the RMSE between the

![Figure 6: Comparison of observed transmission loss with modeled TL: observed TL (black); best model of each independent GA population (gray); maximum a posteriori model (red).](image)

![Figure 7: Comparison of observed transmission loss with modelled TL for frequencies not used in the inversion procedure: observed TL (black); maximum a posteriori model (red).](image)
two curves is indicated. The RMSE varies between 1.5 dB for 1270 Hz and 2.6 dB for 800 Hz. The modelled TL can track relatively well the real data TL, in particular for the highest frequency. Figure 7 shows another validation test, where frequencies up to 8 kHz are included, in order to check the validity of the estimated parameters for a frequency range outside the frequency range used in the inversion procedure. The RMSE estimates increase slightly for frequencies 1600 and 2540 Hz, up to 4.6 dB, but remain bounded to 2.6 dB for frequencies 4032 and 8064 Hz, i.e., a similar amount of error as frequencies up to 1270 Hz.

4. CONCLUSIONS

Acoustic inversion results based on a single hydrophone were obtained for seafloor parameters of a three-layer model. The acoustic inversion procedure was based on the transmission loss measured across a 3 km transect for 7 frequencies in the band 318 to 1270 Hz. Seafloor parameters and source and receiver depths were included in the search space, and the optimisation was carried out by means of a genetic algorithm. Source and receiver depths were estimated with realistic results, were source depth 10.4 m (true value approximately 12 m), and receiver depth was estimated 5 m (true value approximately 8 m). Concerning the seafloor parameters, sound speed in sediment and sub-bottom provided the most compact a posteriori distributions, with credible peak values. A validation step based on the comparison of measured TL with replica TL at frequencies not used in the inversion procedure, in the band 1600 to 8064 Hz, was carried out. While the RMSE for inversion data was in the range from 1.5 to 2.6 dB, for the higher frequencies this indicator ranged from 2.5 dB (at 8054 Hz) to 4.3 dB (at 2540 Hz).

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FIELD DEPLOYMENTS OF A SELF-CONTAINED SUBSEA PLATFORM FOR ACOUSTIC MONITORING OF THE ENVIRONMENT AROUND MARINE RENEWABLE ENERGY STRUCTURES

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\textbf{Abstract:} The drive towards sustainable energy has seen rapid development of marine renewable energy devices, and current efforts are focusing on wave and tidal stream energy. The NERC/Defra collaboration FLOWBEC-4D (Flow, Water column & Benthic Ecology 4D) is addressing the lack of knowledge of the environmental and ecological effects of installing and operating large arrays of these devices. The FLOWBEC sonar platform combines a number of instruments to record information at a range of physical and multi-trophic levels at a resolution of several measurements per second, for durations of 2 weeks to capture an entire spring-neap tidal cycle. An upward-facing multifrequency Simrad EK60 echosounder (38, 120 and 200 kHz) is synchronised with an upward-facing Imagenex Delta T multibeam sonar (120° x 20° beamwidth, 260 kHz) aligned with the tidal flow. An ADV is used for local current measurements and a fluorometer is used to measure chlorophyll (as a proxy for plankton) as well as turbidity. The platform is self-contained, facilitating rapid deployment and recovery in high-energy sites. Five 2-week deployments were completed in 2012 and 2013 at wave and tidal energy sites, both in the presence and absence of renewable energy structures. These surveys were conducted at the European Marine Energy Centre, Orkney (Scotland). Using multifrequency target identification and multibeam target tracking, the depth preference and interactions of birds, fish schools and marine mammals with renewable energy structures can be tracked, together with dive profiles and predator-prey interactions. Measurements from the subsea platform are complemented by 3D hydrodynamic model data, concurrent shore-based marine X-band radar and shore-based seabird observations. These datasets offer insights into how fish, seabirds and marine mammals successfully forage within dynamic marine habitats, how marine energy devices might alter the behaviour of such wildlife and whether individuals face collision risks with tidal stream turbines.
Keywords: Marine renewable energy, environmental monitoring, collision risk, remote sensing, multibeam sonar, echosounder, fish, marine mammals, seabirds, predator-prey.

1. INTRODUCTION

Little is known of the environmental and ecological effects of installing and operating wave and tidal stream marine renewable energy devices (MREDs) [1]. The NERC/Defra collaboration FLOWBEC-4D (Flow, Water column and Benthic Ecology 4-D) is investigating the potential effects of MREDs at test sites in Orkney at the European Marine Energy Centre (EMEC). The project aims to understand how currents, waves and turbulence at wave and tidal energy sites may influence the behaviour of marine wildlife, and how MREDs might alter the behaviour of such wildlife as single devices are scaled up to arrays. Mobile predator and prey use of high-energy sites is being investigated to identify and quantify which type of habitats (depth of water column, speed of tides, etc.) predators predictably use in these areas for foraging, to assess collision risk.

Trends and predator-prey interactions in these sites are known to occur over a variety of temporal and spatial scales [2] requiring data to be captured at a high temporal resolution (several measurements a second) but also for entire spring-neap tidal cycles (2-weeks). Sampling at different positions within these wave and energy sites is also required, to understand the use of habitats by different species and to assess the effect of the presence / absence of MREDs.

Regulators need to know with a high degree of certainty whether tidal and wave devices will affect the population level of marine species, but measuring population level changes is a long term and large spatial range issue. An approach which can rapidly and accurately identify and quantify any changes in individual behaviour, within a species, brought about specifically by renewable development, can allow the quantification of what those impacts will be at the population level [3].

Although boat surveys can provide high-resolution survey coverage along a track, it is not logistically feasible to monitor a high-energy site continuously at high-resolution for an entire 14-day tidal cycle. The effects of wind, waves and tide reduce positional accuracy such that boat surveys are not able to monitor small-scale interactions of individual targets with MREDs, and the cost of long duration surveys is high. Surface moorings, such as instrumented buoys and surface platforms can reduce cost and increase survey duration, but taking high-resolution measurements of the entire water column and measuring the interactions of wildlife with seabed MREDs is limited from an unstable surface platform in these high-energy sites [4]. In the case of both boats and moored surface platforms, there is also the risk of the surface presence affecting the species being studied (e.g. birds, fish and marine mammals) [5, 6].

Mounting instruments on the MRED of interest provides a stable mounting and simplifies power and data requirements. The interactions of fish with tidal turbines have been imaged using cameras but visibility (turbidity and illumination) limits both the range and survey time of a visual approach [7]. Acoustic instruments mounted on the MRED are adversely affected by turbulence from the MRED itself which can mask the presence and interactions of wildlife [8]. An independent platform allows the instruments to be positioned adjacent to the MRED looking at the MRED from a short distance, allowing the interactions of wildlife to be imaged.
but also allowing baseline studies to be conducted under similar conditions in an area free from MREDs. This was the approach chosen for the FLOWBEC project.

2. METHODOLOGY

The FLOWBEC project combines data from the deployment of an upward-facing subsea sonar platform with shore-based bird observations, shore-based marine X-band radar surveys of wave and current data [9] and detailed 3D modelling of the flow and water column.

The FLOWBEC upward-facing sonar platform allows the interaction of fish, diving seabirds and marine mammals with MREDs to be imaged, and the acoustic environment analysed as shown in Fig. 1. The platform combines an Imagenex 837B Delta T multibeam sonar (260 kHz) pinging at several frames per second for target tracking, identification and behavioural analysis, synchronised with a Simrad EK60 multifrequency echosounder (38, 120 and 200 kHz) used for target identification, abundance estimates, measures of plankton and the morphology of turbulence [10].

Fig.1: The multibeam sonar (left) images the water column along the axis of tidal flow for tracking targets and monitoring their interactions with tidal turbine structures. The left image shows a target tracked for 7-seconds at the EMEC tidal energy site, tracked swimming with the tidal flow over 10 m in the vicinity of a turbine structure. The Simrad EK60 multifrequency sounder (right) faces vertically upwards for target identification, abundance estimates and measures of the morphology of turbulence. A 5-minute excerpt of EK60 data at the EMEC wave energy site shows diving seabirds (guillemots / razorbills, confirmed by concurrent shore-based observations) feeding on the fish shoal.

The multibeam sonar (MBES) is aligned with its 120° swath orientated parallel to the tidal flow, and inclined so that the outer beams are parallel to the seabed to include the MRED within the swath. Using this orientation, diving seabirds can be detected above water by the shore-based radar and bird observer, before being tracked underwater in the tidal flow, and the interactions of fish, seabirds and marine mammals with the MRED can be monitored.

The Delta T MBES images a wide swath of 120° by 20°, with 120, 240 or 480 beams, 500 range bins and at repetition rates of up to 20 pings/second. Its range can be adjusted from 0.5 to 100 m and all parameters (range, gain and ping scheduling) can be controlled programmatically via TCP commands in real-time during operation. The Delta T MBES was selected for its low cost and power consumption (typically < 10W), with similar Imagenex
Delta T models having already been used successfully for a variety of applications [11]. The MBES measures the backscattering strengths (in dB) of all targets, relative to a source level of 190 dB re. 1 µPa @ 1 m (Patterson, pers. comm., 2012). Pulse lengths vary with the range setting (e.g. 300 µs at 50 m range).

The three frequencies of the EK60 echosounder (38, 120 and 200 kHz) have 7° conical beams orientated vertically upwards, facing the water surface. Comparison of scattering strengths at the different frequencies enables identification of fish species, and this echosounder has also been used successfully to examine diving seabirds (e.g. [12]).

An Acoustic Doppler Velocimeter (ADV) provides local current measurements and a fluorometer is used to measure chlorophyll (as a proxy for plankton) and turbidity.

Onboard batteries and data storage for two-week deployments allow an entire spring-neap tidal cycle to be captured. The self-contained seabed platform can be positioned close to the MRED to be investigated allowing the interactions of wildlife to be imaged, but also allowing baseline studies to be conducted under similar conditions in an area free from MREDs or prior to MRED installation.

3. RESULTS

Five 2-week deployments have been completed at wave and tidal energy sites at EMEC in Orkney (UK). Deployments were conducted during the seabird breeding season which peaks in summer months [13]. Deployments adjacent to MRE structures (10-20m vicinity) and at ‘control’ sites in areas free from MREDs were carried out to assess the effect of the presence / absence of these devices.

Algorithms for noise removal, target detection and tracking have been written. Fig. 2 shows an example fish shoal tracked using the multibeam within a few metres of the Atlantis AK-1000 tidal turbine structure (shaded in green) at the EMEC tidal site. The turbine blades were not present during this deployment and their expected radius is outlined with a dashed green line.

Fig.2: The multibeam swath shows a large fish shoal, tracked over 12-seconds. The turbine structure is outlined in green. The turbine blades were not fitted but their expected radius is outlined in dashed-green. The same target is detectable using the EK60 (shown on the right), where the frequency response can be used to aid target identification [14].
It is possible to estimate collision risk and the impact of the turbine structure on tracked targets by considering the vertical distribution of targets and what percentage are likely to encounter the turbine structure. Fig. 3 shows the vertical distribution of all tracked targets for a 14-day survey adjacent to a tidal turbine structure. The water column up to a height of 21 m above the seabed is shown. The turbine structure is shaded in green (to scale) and the dashed outline indicates the expected radius of the blades (blades not present during this survey).

Fig. 3: The vertical distribution of tracked targets in the lower 21 m of the water column over a 14-day survey adjacent to a MRE structure. Many targets are at a height where they are likely to encounter the turbine structure (or blades when present).

4. DISCUSSION

The next step is target classification to guide species identification and to allow analysis by time, tide and space for a specific category of targets. Target classification is possible using a variety of methods. The morphology (size, shape, intensity, number of targets per frame, target separation) and behaviour (velocity, velocity relative to water column, directionality, vertical distribution and inter-target interaction) can be observed using the multibeam, and classification performed by defining ranges for the various parameters.

Fig. 4 shows four example raw multibeam scans from a deployment adjacent to a tidal turbine structure showing a large shoal of fish, an individual target, a smaller, more densely packed shoal of fish (possibly a different species) and a diving seabird.
Fig. 4: Four examples of different target types in the vicinity of a tidal turbine structure shown from raw 40 m range multibeam data averaged over several seconds. The turbine structure is shaded in green, and the expected blade radius is outlined by a dashed green line (blades not present for this survey). Image A shows a large shoal of fish, B shows a small individual target passing through the midwater, C shows a denser shoal of fish (possibly a different species) and D shows a diving seabird.

Target classification is also possible using multifrequency analysis from the EK60 echosounder data [14]. For fish, the known frequency response of different fish species can be used to identify pelagic and demersal species, and to train software to pick out and track a range of different shoaling / feeding behaviours using the EK60 for identification and the MBES for tracking. The fish shoal in Fig. 2 was also shown in the EK60 echogram for each of the three frequencies.

The shore-based wildlife observations are used for ground truthing, particularly for identifying seabird species on the multibeam by their distinctive dive behaviour. A subset of shore-based bird observations can be used to first ground-truth acoustic detection of diving seabirds in both sonar instruments, and second to use the known identification of species to ‘train’ software to pick out different species. The software can then be tested with the remaining shore-based observations.

The outcome of the tracking analysis will allow the environmental effect of MREDs to be explored using the distribution of targets (plankton, fish, birds, marine mammals) and predator-prey interactions with time, tide and space, where space includes vertical use of the water column, and horizontal distribution around the wave and tidal sites, and how all of this changes with the presence and absence of MREDs. The vertical habitat preferences of these ecological groups and collision risks can also be evaluated by looking at spatial overlap with MREDs, and collision risk predicted by looking at the overlap with conditions favoured for MREDs.
5. CONCLUSIONS

Increasing commitments to renewable energy in short timescales have seen rapid development of marine renewable energy sources and devices. Little is known of the general effects of installing and operating MREDs, at all depths and in all environments. The FLOWBEC project aims to address the challenge of monitoring a significant portion of animal activity, biological and physical dynamics within the water column and at the sea surface near MREDs, using below-the-water instruments like sonars and above-the-water sensors like radar.

The technology and analytical approach developed in FLOWBEC are currently the only subsurface system to continuously capture fine-scale (several measurements a second, sub-metric spatial resolution) data over a wide range of both physical and multi-trophic levels (plankton, zooplankton, fish, seabirds and mammals) over time periods which encompass day and night differences as well as full spring / neap tidal cycles.

The Imagenex 837B Delta T multibeam sonar provides high resolution information on a variety of targets in the water column around MREDs. The combined use of an EK60 multifrequency echosounder enables the identification of fish species and has the potential for the identification of seabirds and marine mammals. Fish, marine mammals and diving seabirds can all be tracked during their interactions with MREDs, above water and below water. Acoustic measurements are being analysed as a function of time, tide, waves, modelled data and shore-based wildlife observations and marine X-band radar to understand the hydrodynamic habitat preference of various functional ecological groups (benthos, plankton, fish, birds and marine mammals) and how individual species may use preferred flow conditions.

Techniques for analysing the raw data and statistical modelling are being refined, such that the combination of the technology and the analysis will ultimately provide an affordable way to measure interactions of marine wildlife in high-energy locations and around MREDs. This combination of our current technology and analytical approach can help to de-risk the licensing process by providing a higher level of certainty about the behaviour of a range of mobile marine species in high-energy environments.

It is likely that this approach will lead to greater mechanistic understanding of how and why mobile predators use these high-energy areas for foraging. If a fuller understanding and quantification can be achieved at single demonstration scales, then the predictive power of the outcomes might lead to a wider strategic approach to monitoring and possibly lead to a reduction in the level of monitoring required at each commercial site.

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UNDERWATER NOISE EMISSIONS FROM DRILLSHIPS IN THE ARCTIC

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Abstract: In 2012, Shell used two drillships to drill top holes of exploratory wells offshore Alaska—one in the Chukchi Sea and one in the Beaufort Sea. A dedicated acoustic monitoring program was performed to measure sounds as a function of range and direction from each of the drillships and support vessels during different stages of the operations. Sounds from the drillships were characterized while a mud line cellar was excavated and while drilling. Broadband (10 Hz-32 kHz) sound levels measured during mud line cellar excavation exceeded those emitted during drilling by 10-20 dB. The spectra for the two activities differed most at frequencies greater than 100 Hz. Estimated broadband source levels for the two measured drill rigs were 193 and 192 dB re 1 µPa during mud line cellar excavation and 182 and 173 dB re 1 µPa while drilling.

Keywords: drillship, floating drill rig, underwater noise emissions, Arctic offshore drilling
1. INTRODUCTION

Underwater sound levels were measured during Shell’s (Shell Offshore, Inc., and Shell Gulf of Mexico, Inc.) drilling program offshore Alaska in summer 2012. Exploratory wells were drilled from the drillship Noble Discoverer (Discoverer) in the Chukchi Sea and the drillship Kulluk in the Beaufort Sea. Resupply, refuelling, anchor handling, oil spill contingency response, and ice management were provided by 15 support vessels.

The drillships and support vessels generated underwater sound levels that could potentially disturb marine mammal behaviour and distribution, or temporarily or permanently lower marine mammal hearing thresholds. As such, and in accordance with United States (US) National Marine Fisheries Service and US Fish and Wildlife Service requirements, Shell implemented a marine mammal monitoring and mitigation program to avoid, where possible, or minimize effects to marine mammals from the drilling activities.

The monitoring and mitigation program included a dedicated acoustic measurement study to characterise sound levels emitted by each drillship and by associated support vessels and ancillary activities. The measurements were combined with marine mammal observational data for post-season estimates of the industrial sound levels to which marine mammals were potentially exposed during the program. The measurements also formed a dataset for assessment of Shell’s future offshore drilling activities in Alaska.

Arrays of autonomous acoustic recorders (AMARs, JASCO Applied Sciences Ltd.) recorded underwater sound levels as a function of range and direction from the noise sources [1]. Four acoustic recorders were deployed at ranges between 1 and 8 km from each drillship. One remote telemetry buoy was deployed in the Beaufort Sea 500 m from the Kulluk to monitor drilling sounds in real time. Sounds from transiting support vessels were measured by three separate acoustic recorders. The vessels ranged in size from a 37 m tug to a 228 m fuel supply vessel. These vessel data are not included in this summary report.

In addition to the recorders used to characterise sound sources, large scale acoustic arrays also collected data at long ranges [2,3] using AMARs in the Chukchi Sea and DASARs (Greeneridge Sciences) in the Beaufort Sea. The primary purpose of the large-scale arrays was to determine presence and distribution of marine mammals by analysing vocalization detections, but the data were also used to examine low-level, long-range sound propagation from the drilling activities. Data from the large-scale arrays are not presented here.

This paper contains the measured rms (root-mean-square) sound pressure levels (SPL) received at each recorder during drilling and mud line cellar excavation. Spectral data and estimated source levels for each drill rig are provided.
2. METHODS

1.1. Data Acquisition

AMARs were deployed along lines that extended radially from each drillsite with nearly constant water depth at each recorder (Fig. 1). The water depth at the Chukchi Sea site was 45 m and that in the Beaufort Sea was 33 m. The recorders in the Beaufort Sea were set along a radial oriented southwest from the drillsite, an orientation that was selected so the recorders would be on the side of the rig closest to the noise-emitting power generators. Those in the Chukchi Sea were set along a radial oriented northeast from the drillsite, to measure sounds along the principle axis of the bowhead whale migration path. This was done rather than aligning the recorders with a particular aspect of the Discoverer because the Discoverer could change its orientation.

![Fig. 1: Drillsites and recorder deployment locations in the (left) Chukchi Sea and (right) Beaufort Sea.](image)

1.2. Sound sources

The Discoverer is a 157 m drillship with an 8.2 m draft. This is a turret moored vessel, which means its mooring anchors are connected to a centrally located turret, about which the vessel can rotate in order to advantageously orient itself in inclement weather. The Kulluk is a floating conical drilling unit 81 m in diameter with a 5.0 m draft.

During drilling, noise from equipment and machinery on board the drill rigs was expected to dominate the total noise emissions. The generators, engines, pumps, hydraulic units, motors, mixers, etc. were located mainly below deck, interior to the hull of the ships, well coupled to the water, which facilitated noise transmission and vibration into the surrounding water.

Part of the drilling process included the excavation of a mud line cellar (MLC)—a pit 6 m in diameter and 12 m deep—constructed to ensure the wellhead and blow-out preventer would be located safely beneath the maximum depth for ice-keel gouge. The bit used to excavate the MLC consisted of series of 0.9 m diameter disks, powered by three
hydraulic motors, and angled to displace seafloor sediments. Noises from this activity were considered separately from the actual drilling.

1.3. Data Analysis

Throughout operations, activity logs from each vessel were maintained by on board Protected Species Observers (PSOs). Their logs were used to identify analysis time windows for the activities under consideration.

Acoustic data were analysed with custom processing software that output spectral and broadband sound pressure levels (SPLs) in specified time windows. SPLs were calculated with Hanning-weighted time windows, 60 s in length, with 50% overlap. Ten minutes of data (from 20 consecutive time windows) were averaged, by frequency, to obtain spectra for sounds from drilling and from MLC excavation. The spectra were then integrated into 1/3-octave bands.

Source sound levels, in 1/3-octave bands, were derived by applying a back-propagation correction at each frequency to the data measured at the closest range. Sound levels were back propagated using transmission loss values calculated from a numerical sound propagation model. The numerical model fully accounted for both seafloor and water surface reflections as well as the acoustic properties of the water column and the seafloor. The model was used to compute sound transmission loss in 1/3-octave bands.

Transmission losses for frequencies below 2 kHz were calculated using the wide-angled parabolic equation model RAM [4], adapted to account for shear wave loss through a complex density approximation. Frequencies 2 kHz and greater were modelled using the ray tracing code Bellhop [5]. In each case, the model input parameters included the geoacoustic properties of the sub-bottom, as well as a definition of the bathymetry at the site and of the sound speed in the water column as a function of depth (Annex A2). The water depth was assumed to be a constant 48 m in the Chukchi Sea and 33 m in the Beaufort Sea. Sound speed profiles for the water column were obtained from temperature and salinity profiles collected at the study sites during the acoustic monitoring program.

3. RESULTS

Spectra for data collected during drilling at 1 km from the Discoverer contained tones around 54, 80, and 100 Hz plus two strong tones at 256 and 300 Hz, each with higher harmonics present up to 10 kHz (Fig. 2, left). The spectrum measured when the Kulluk was drilling also contained several tones below 200 Hz and a strong tone at 256 Hz with higher harmonics up to 10 kHz (Fig. 2, right). The lower frequency tones are likely associated with power generation equipment and motors on the rigs. The higher-frequency tones could be from the shakers and agitators of the mud systems on the rigs.

Fig. 3 is a plot of the power spectral density measured during drilling at ranges between 1 and 8 km for each drill rig. The power of the high frequency harmonics decayed with range due to absorption loss at the high frequencies. 1/3-octave band source level estimates for each drill rig while drilling are shown in Fig. 4 along with those for mud line cellar excavation. These levels resulted in broadband source level estimates of 182 and 173 dB re 1 μPa for the Discoverer and the Kulluk, respectively, while drilling.

The received sound levels were elevated when each rig was excavating the MLC, over those received during drilling, particularly at frequencies greater than 100 Hz. During MLC excavation, the high frequency tones and harmonics were obscured by sound with a
more broadband nature, a characteristic that could indicate these sounds originate from the MLC bit grinding into the seafloor or from sounds from the MLC bit itself. When the mud line cellar was being excavated, these broadband source level estimates were 193 and 192 dB re 1 µPa for the Discoverer and the Kulluk, respectively.

![Fig. 2: Power spectral density measured at 1 km range for MLC excavation and for drilling (left) by the Discoverer and (right) by the Kulluk.](image1)

![Fig. 3: Power spectral density measured at different ranges for drilling (left) by the Discoverer and (right) by the Kulluk.](image2)

![Fig. 4: 1/3-octave band source level for (left) the Discoverer and (right) the Kulluk during drilling.](image3)
4. CONCLUSION

Underwater sound levels were recorded during drilling at two locations offshore Alaska in the summer of 2012. Sounds during mud line cellar excavation exceeded those emitted during drilling by 10-20 dB. The estimated broadband underwater source level for the drillship Noble Discoverer was 182 dB re 1 µPa during drilling, 193 during mud line cellar excavation. Broadband source levels for the Kulluk were estimated to be 173 dB re 1 µPa while drilling and 192 dB re 1 µPa while excavating the mud line cellar. During drilling each drill rig emitted tones that could be associated with power-generating equipment or other equipment on the rigs. At frequencies above 100 Hz, these tones were obscured by sound more broadband in nature when the mud line cellar was being excavated.

5. ACKNOWLEDGEMENTS

These results are a culmination of the skilled and dedicated work of the field teams and data analysts from JASCO Applied Sciences, particularly Andrew McCrodan, Caitlin O’Neill, Alex MacGillivray, and Zizheng Li. We gratefully acknowledge the assistance from the crews on the vessels MSV Fennica and MSV Nordica. Shell funded this work.

REFERENCES


**ANNEXE: PARAMETERS USED TO BACK-PROPAGATE DRILLING SOUNDS**

<table>
<thead>
<tr>
<th>Depth (m)</th>
<th>Density (g/cm³)</th>
<th>Compressional Wave Speed (m/s)</th>
<th>Compressional Wave Attenuation (dB/λ)</th>
<th>Shear Wave Speed (m/s)</th>
<th>Shear Wave Attenuation (dB/λ)</th>
</tr>
</thead>
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<tr>
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<td>1.83-1.89</td>
<td>1701-1763</td>
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<td></td>
</tr>
<tr>
<td>70-170</td>
<td>1.99-2.12</td>
<td>1813-1927</td>
<td>0.2</td>
<td>113</td>
<td>1.7</td>
</tr>
<tr>
<td>&gt;170</td>
<td>2.12</td>
<td>1927</td>
<td>0.2</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Table 1: Geoacoustic parameters input to the numerical model used to back-propagate the received sound levels from drilling in the Chukchi Sea.

<table>
<thead>
<tr>
<th>Depth (m)</th>
<th>Density (g/cm³)</th>
<th>Compressional Wave Speed (m/s)</th>
<th>Compressional Wave Attenuation (dB/λ)</th>
<th>Shear Wave Speed (m/s)</th>
<th>Shear Wave Attenuation (dB/λ)</th>
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<td>2.2-2.4</td>
<td>1674-1702</td>
<td>0.3-0.2</td>
<td>200</td>
<td>2.6</td>
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<tr>
<td>30-200</td>
<td>2.4</td>
<td>1673-1843</td>
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<tr>
<td>&gt;200</td>
<td>2.4</td>
<td>1843</td>
<td>0.2</td>
<td></td>
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</tr>
</tbody>
</table>

Table 2: Geoacoustic parameters input to the numerical model used to back-propagate the received sound levels from drilling in the Beaufort Sea.

**Fig. 5: Water column sound speed as a function of water depth, derived from temperature and salinity profiles measured at Burger and at Sivulliq.**
Abstract: In the oil & gas industry there is a trend towards more subsea activities. To improve gas recovery from existing and new fields at greater depths, the produced gas will be compressed, processed and transported via subsea templates and underwater networks (pipelines, flexible risers, etc.). Besides the huge consequences for the subsea installation itself (reliability, maintenance, etc.), it also has consequences for underwater wildlife through the underwater source vibrations leading to sound radiation. Until now very little is known about the underwater source mechanisms, the acoustic strength of these underwater networks, the coupling of the emitted source sound to the surrounding medium and the impact of the sound on the underwater wildlife. This paper presents a simplified model for a subsea high speed turbo-compressor coupled to the KrakenC normal mode propagation model. With this combined model the noise at remote locations can be predicted and compared with the ambient noise and other anthropogenic noise sources such as for instance shipping, dredging and wind farm operation noise.

Keywords: High speed turbo compressor, underwater source level, environmental impact, modelling, sound propagation, measurements

1. INTRODUCTION

The increasing demand, technological developments and decreasing production in shallow water fields in the last decades has led to considerable investments and improvements on oil & gas field deployments at deep and also ultra-deep waters. Whereas in many cases these fields are unprofitable to operate with conventional, expensive (floating) production platforms, the above mentioned arguments are leading to the development of subsea networks, ranging from relatively straightforward manifolds to complete, complex production installations. In the near future these production installations will also include subsea compression, which is a major technological leap forward. Such developments have high demands and consequences on the robustness
(reliability, maintenance, inspection, etc.) of the installation itself throughout the whole life of the installation and the field. Besides that, especially subsea compression with high speed and high pressure turbo-compressors can produce a considerable amount of noise. Up to now, little is known about the subsea noise contribution of these production installations. An important reason is that no noise measurements are available yet and the noise radiation from a vibrating structure in water differs considerably from air. Moreover, in contrast to most other rotating equipment such as pumps, high speed and high gas flow turbo-compressors installations produce relatively high frequency noise. In [1], the underwater source level was estimated using the measured in air spectrum, correcting for the effect of fluid loading on the radiation efficiency. In this paper the turbo-compressor source level is put into perspective by comparing it against the source level of shipping, dredging, wind- and tidal turbine levels. Also, predicted compressor noise levels are compared against typical hearing thresholds of fish and mammals for a typical deep and shallow north sea environment.

2. TURBO-COMPRESSOR SOURCE LEVEL AND CONSIDERED SCENARIOS

With the measured noise spectrum (in air) of a complete turbo compressor installation and the known differences between radiation into air and sea water, a stylized, equivalent noise source power spectrum was constructed in [1] (figure 1, black line). To put the magnitude of the source level of the compressor in perspective, it is compared with the monopole source level of shipping [2] (red), tidal turbines [3] (magenta), wind farms (blue) and dredging activities [4] (green). The wind farm source level plateau was determined by taking the maximum spectral density level of various measured operational source levels as compiled in [5, figure 5(a)]. The estimated source spectrum of the turbo-compressor suggest that the noise at frequencies larger than 1 kHz exceed typical levels of noise produced by shipping and dredging activities by 10 – 20 dB, whereas for lower frequencies < 1 kHz, it is much more quiet and more comparable to low noise levels generated by wind and tidal turbines during operation. The 95% certainty interval plotted for the turbo-compressor is the uncertainty in the computed radiation efficiency as a function of the modal density and the internal dissipation.

![Figure 1: Example of the estimated monopole source spectral density of a turbo-compressor (black), with other sources superimposed for comparison: shipping (red), dredging noise (upper limit, green; lower limit green dashed), wind farm operation (blue) and tidal turbine inflow turbulence noise (magenta).](image-url)
3. MODELLLED TURBO-COMPRESSOR NOISE LEVELS

The turbo-compressor source spectrum is used to compute noise levels for two typical north sea environments: a remote location in shallow waters in the Dutch sector of the North Sea and a deeper location in the Skagerrak trench. We consider two different scenarios: a shallow case (50 meter depth) and a deep case (400 meter depth). Typical sound speed profiles (SSPs) were obtained from the World Ocean Atlas database to assess the effect of seasonal variation in predicted noise levels. Surface losses due to wind were taken into account by modelling a surface roughness representing a typical 6 m/s wind speed at 10 m above the sea surface. The sediment was modelled as being medium sand. The source was modelled as a point source located 2 meter above the sea bottom. Volume attenuation was modelled using the ‘Thorp’ equation. Next, the KrakenC [6] propagation model was used to compute the contribution of the turbo-compressor radiated power to the underwater noise. The normal mode modelling approach is a computationally efficient method for computing the propagation loss in shallow and medium constant depth water.

Figure 2: comparison of depth averaged one third octave band levels for the shallow (50m) and deep water (400m) scenario at 1 km, 25 km and 100 km distance from the turbo-compressor. The solid and dashed lines indicate the energetic mean computed using SSPs for each month. The uncertainty bounds are indicating the maximum and minimum computed SPL. The predicted uncertainty of the source level (figure 1) has not been included in the uncertainty bounds.

Figure 2 illustrates the computed one third octave SPL for the shallow and deep water scenario. The solid and dashed line indicate the energetic mean of the 12 months. The uncertainty bounds indicate the minimum and maximum encountered SPL. The large spread is explained by the variation of the sound speed profile throughout the year. The large spread at higher frequencies for the deep water scenario is due to scattering losses at the surface and the upward refracting sound speed profile common during the winter months. Due the low energy radiated by the source at lower frequencies, there is little contribution to the low frequency noise levels around the compressor. In addition, due to attenuation and scattering from the bottom and surface, the very high frequencies are attenuated relatively quickly. The ‘mid-frequency’ bands (~1 - 4 kHz) however contain most of the acoustic energy and are attenuated slowly, resulting in...
long distance propagation. It should be noted that in this frequency range, sea surface scattering is dominated by the presence of wind-generated bubbles [7], which were not modelled in Kraken.

4. ENVIRONMENTAL CONSIDERATIONS

Fish and marine mammals hear well underwater, and rely on sound for instance to communicate, orientate and listen for predators. The potential impact of introducing these new turbo-compressors into the subsea environment will strongly depend on the type of taxa/species considered. Here we investigate the audibility of the noise generated by comparing the estimated noise levels to audiograms reported in the literature. Figure 3 compares the predicted third octave band levels for the turbo-compressor to audiograms of different fish (top) and marine mammal species (bottom).

Background noise can also mask animal hearing, which can reduce the distance to which animals can perceive sound in practice. We superimpose a typical background noise curve in the form of the Wenz curve for heavy shipping intensity and wind speeds of 9 m/s. To allow a comparison to the predicted noise levels, we add a critical ratio (CR) that accounts for the integration bandwidth of the animal hearing. For the purpose of this work, we adopt a generalized CR dependency on frequency for fish and marine mammal species. For fish, we base this on an extrapolation based on cod data used in [8]: CR(f) = 10 log₁₀ (f/2), in dB re 1 Hz. For the marine mammals we adopt the CR relation as measured by [9] for the harbour porpoise: for f < 4 kHz CR(f) = 18.3 dB re 1 Hz, and for f >= 4 kHz, CR(f) = 10.7 + 12.1·log₁₀ (f/1000) dB re 1 Hz, with f the frequency in Hz.

The audiograms in figure 3 provide insight into what distances sound can be perceived by different species. From figure 3 it can be expected that non-hearing specialist fish (such as salmon and cod) will perceive the compressor sound only at relatively close distances (< few km) to the source. For hearing specialists that can perceive higher frequency sounds, such as the herring, the predicted noise levels drop below the audiogram at distances of ~ 25 km (depending on propagation conditions). Although some species (such as the catfish) have sensitive hearing at higher frequencies, the ambient noise is likely to mask the compressor noise at distances of 100 km. The interpretation of measured audiograms for fish are still under debate. Audiograms are commonly measured in tanks, where the relation between particle motion and sound pressure is different from the natural environment, and only few experiments exists that address this issue (see [10]).

Marine mammals are typically grouped into different hearing categories: low, mid, high frequency cetaceans, and pinnipeds [e.g. 11]. Comparing the predicted turbo-compressor noise to audiogram limited and background limited thresholds in figure 3 suggests that belugas may perceive the sound at distances of ~ 25 km, whereas porpoises and seals may even perceive the sound up to 100 km distance, but only in good propagation conditions. No measured audiograms are currently available for low frequency cetaceans. Although theoretical considerations indicate that minke and humpback whales are sensitive to sound between 1 and 10 kHz [12][13], absolute hearing thresholds are still lacking. Recent studies of responses of baleen whales to sonar show that low frequency cetaceans respond to sound in the mid-frequency (1 – 10 kHz) range [14,15], albeit at much higher levels than those predicted for the turbo-compressor sounds.

5. CONCLUSIONS AND RECOMMENDATIONS

The subsea turbo-compressor source model was estimated in [1] using in-air measurements. While shipping, dredging and wind farm operation contain their dominant noise contribution at low frequencies, it was found that the turbo-compressor has a significant noise component in the
Figure 3: comparison of depth average one third octave levels at different distance from the turbo-compressor (figure 2: magenta (1km), red(25km), blue(100km)) with the hearing thresholds of a variety of fish and mammals. Data origin: fish curves, Atlantic Herring, Atlantic cod, Atlantic salmon, and channel catfish (from [8][10], and references therein), harbour porpoise ([16]), harbour Seal ([17]), Beluga (composite of [18] and [19]). The green curves indicate generalized masked detection thresholds, NL(f)+CR(f), using two approaches for fish and marine mammals (see text for details).

mid-frequency (1 – 4 kHz) range. As there are a lot of assumptions required to obtain an underwater monopole power spectrum, measurements should be carried out to verify the model. It is planned to carry out such measurements in the near future.

The noise contribution in one third octaves was modelled for a typical shallow (50 m depth) and deep water (400 m depth) North Sea environment. Simulations showed that predicted noise levels are strongly seasonally dependent. Scattering has an important effect on the propagation loss at higher frequencies, especially for the deep water scenario where the cooling of the ocean surface resulted in an upward refracting sound speed profile during the winter months.

By comparing the predicted noise levels to audiograms for different marine mammal and fish species, we show that there is a large between species spread in distances to which the turbo-compressor sound can be perceived, which can range from ~ a few km for some fish species, out to ~100 km for some cetaceans in good propagation conditions. Given the overlap in frequency band containing the dominant component of the turbo-compressor and hearing ranges of marine mammals, and some fish, the potential of disturbance or masking should be considered when introducing this new type of source in the underwater environment.
REFERENCES

NEW METHODS IN IMPACT PILE DRIVING NOISE ATTENUATION

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Abstract: Underwater noise from impact piling reaches peak sound pressure levels on the order of ~103 Pa at range 3000 m, ~104 Pa at range 60 m and ~105 Pa at range 10 m. These peak pressures have deleterious effects on underwater fauna resulting in regulatory agencies placing limitations on offshore impact pile driving.

In this paper, we review current noise attenuation methods and discuss their limited success. Our inspection of the mechanics behind impact pile driving finds the rise time in the pile’s propagating stress wave responsible for peak acoustic pressures. This knowledge guides us to alternate methods to achieve noise reduction.

Of the possible methods, two stand out as economically feasible, structurally sound and high implementable. It is these two methods, the double pile and the zero Poisson’s ratio (ZPR) pile, that are presented in detail. Both finite element analysis and sub-scale testing promise greater than a 20 dB reduction in peak sound pressure levels and full scale testing is in preparation. Noise reduction at these levels will have a profound positive effect on the industry of marine piling.

Keywords: Double pile, Impact, Marine, Piling, Sound pressure level, Underwater noise, Zero Poisson’s ratio (ZPR)
1. INTRODUCTION

Impact pile driving has been associated with extremely high sound pressure levels [1]. Underwater anthropogenic noise has widespread effects on marine fauna [2]. In the Northwest region of the United States much of the marine fauna being affected is listed under the Endangered Species Act (ESA) or the Marine Mammals Protection Act (MMA). Regulatory agencies in the United States; National Oceanic and Atmospheric Administration (NOAA) and the United States Fish and Wildlife Service (USFWS), have instituted underwater noise regulations to ensure these species are protected. Southall et al. provides detailed explanation and background surrounding the development of these noise limits [3].

In order to meet these underwater noise regulations many organisations are conducting research to understand pile driving noise and its propagation [4, 5, 6, 7]. A brief review of this work is shown. From the review, two new methods of attenuation are introduced.

2. NOISE GENERATED BY IMPACT PILE DRIVING

Underwater noise is typically described by three metrics and expressed in decibels (dB); peak sound pressure level (Lpeak), root mean square sound pressure level (RMS), and sound exposure level (SEL) [1]. After reviewing each of the metrics it can be seen that a reduction in Lpeak results in a reduction in the other metrics. Therefore this discussion concentrates on the reduction of Lpeak.

Sound pressure associated with pile driving is created by a propagating elastic compression wave that travels down the pile from the pile head to the pile toe. This compression wave is set up by the interaction between the pile driving hammer and the pile head. Associated with the compression wave is local pile expansion. The expansion, or diameter increase, is related to the Poisson’s ratio of the pile material. This expansion of the pile displaces the local fluid and creates an acoustic wave front which then propagates downward and away from the pile.

As the compression wave reaches the pile toe it is reflected and begins propagating upward. The acoustic wave front associated with the upward traveling elastic wave propagates upward and away from the pile. A detailed explanation of this phenomenon can be found in [4]. Through axisymmetric finite element analysis, we also show this phenomenon, see Fig. 1.

Peak sound pressure at the pile wall is directly related to the velocity of pile expansion. This velocity can be found from hammer energy, hammer geometry, pile geometry, and pile material properties.
3. ANALYSIS OF CURRENT NOISE ATTENUATION METHODS

Several authors have summarized the current research in pile driving attenuation [8, 9]. They have outlined the effectiveness of bubble curtains, cofferdams, isolation casings, encapsulated bubbles, pile cushions, soft start methods, and others. At first glance it is easy to see that there is wide variation in the effectiveness among the different methods. However, as a general rule, sound reductions greater than 10 dB cannot be reliably predicted [9].

3.1. Water column attenuation methods

As shown in [4], even perfect attenuation systems that only occupy the water column fail to capture the upward traveling acoustic wave fronts. Although these wave fronts are attenuated by the soil surrounding the pile, partially reflected at the soil-water interface, and cylindrically spread until they reach the hydrophone, they still contain a significant amount of energy. The amount of energy is heavily dependent on the soil attenuation properties. At farther distances from the pile, changes in acoustic energy are attributed to bathymetry, bottom reflection loss, and additional spreading.

3.2. Hammer adjustment methods

The use of pile cushions to reduce acoustic $L_{\text{peak}}$ is known in industry. Pile cushions reduce the rise time in pile stress by delaying the onset of force into the pile. This reduced rise time results in reduced fluid particle velocities, and reduced $L_{\text{peak}}$. The reduced rise time can also reduce pile driving efficiency, possibly increasing the acoustic SEL.
The soft start method also reduces pile noise. To practice the soft start method a pile is initially driven with low hammer energy. As the pile is driven further into the soil, the hammer energy is increased as necessary to achieve soil penetration. The soft start method is intended to be a warning mechanism for fauna so that they can vacate the area before maximum hammer energy is reached.

4.3. Summary

Water column attenuation methods are dependent on both the effectiveness of the attenuation system and on the soil properties near the pile. Hammer adjustment methods may not reduce noise through the entire pile driving cycle and may increase acoustic SEL.

5. NEW ATTENUATION METHODS

Due to the ineffectiveness in existing attenuation methods two new methods are suggested. One is to divorce the pile to water interaction along the entire length of the pile, a double pile. The other is to create a pile that does not displace radially when compressed, a zero Poisson’s ratio pile.

5.1. Double Pile

A double pile is constructed by inserting one smaller diameter pile inside another larger diameter pile, such that the two piles are concentric to one another. The volume between the two piles is sealed from water intrusion and thus full of air. At one end, the pile toe, the two piles are joined to each with a special cutting shoe and flexible connection. This connection is designed such that full force of the hammer from each pile strike is available for driving the pile toe into the soil. As the inner pile toe moves downward into the soil, force is simultaneously applied to the toe of the outer pile thus pulling it into the soil. The force applied to the outer pile is done so through the flexible connection which reduces the rise time of the stress state in the outer pile and acoustic Lpeak created. A pile of this design eliminates all inconsistencies in noise attenuation performance and after pile installation the inner pile may or may not be removed and reused.
Results from finite element analysis and subscale testing are provided in Fig. 4 and Fig. 5 respectively. Both methods suggest a 20dB reduction in peak sound pressure levels.

Fig.4: Acoustic radiation from an axisymmetric finite element double pile model.
5.2. Zero Poisson’s Ratio Pile

One method for constructing a ZPR pile is to use composite fiber winding techniques. This construction technique allows the angle of the composite fibers to be adjusted on a ply by ply basis. The resulting structure has a modulus and Poisson’s ratio that can be adjusted in each of the three primary directions; x, y, z.

Both closed form and finite element methods can be used to determine the fiber angle necessary in each ply to create a ZPR structure. Fig. 6, shows a finite element simulation of a ZPR pile constructed using these methods. Again the analysis shows a 20dB reduction in L\text{peak}.

5.3. Summary

The double pile and ZPR pile eliminate the pile wall as a noise source. The remaining noise source in the system is related to the pile toe advancing downward into the soil. The movement of the pile toe cannot be eliminated, thus the associated noise must remain.
However, by using soft start methods the $L_{\text{peak}}$ generated by the pile toe can be regulated. As the pile penetrates the soil, the acoustic waves generated at the pile toe are attenuated by more and more soil, helping to offset the increasing hammer energy necessary for soil penetration.

It should also be noted that, as in [4], the finite element analysis shown models the soil as a fluid with equivalent soil wave speed. Modelled as such, the soil does not resist pile penetration. This results in artificially high pile toe velocities, hence higher acoustic pressure.

6. CONCLUSION

Current noise attenuation methods have been reviewed and analysed. The source of their ineffectiveness was highlighted.

Two new pile systems were introduced. Both systems eliminate the pile wall as a sound source. The remaining noise generated by the two systems can be regulated by monitoring hammer energy. The key advantage to these pile systems is that the noise created by each system will be consistent from day to day and site to site.

7. ACKNOWLEDGEMENTS

The authors would like to express gratitude to the Washington State Department of Transportation and the program manager, Rhonda Brooks.

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SOIL VIBRATION DUE TO OFFSHORE PILE DRIVING AND
INDUCED UNDERWATER NOISE

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Abstract: Measured data of soil vibration and hydro sound pressure due to offshore pile
driving is evaluated to follow up the secondary transmission path of body and surface
waves from the pile through the soil into the water. The wave propagation by the
secondary transmission resulting in underwater noise circumvents noise mitigation
techniques, which are based on pile-surrounding barriers like bubble curtains. The
attenuation of hydro sound immission is regulated by the German government with
limiting values. Therefore, a better understanding of the whole physical process of wave
propagation due to impact pile driving is necessary. Data of triaxial soil vibration and
hydro sound pressure are evaluated to estimate a possible influence of the secondary
transmission path on underwater noise.

Keywords: Impact pile driving, wave propagation, hydro sound immission, marine
mammals, measurements, soil vibration velocity
INTRODUCTION

For the protection of marine mammals hydro sound attenuation is frequently discussed in Germany due to the aimed expansion of offshore wind energy in the North Sea. Impact pile driving leads to considerable hydro sound pressure levels which disturb or can even harm the harbor porpoise. The German Standards on underwater noise demand a limit of the sound exposure level (SEL) of no more than 160 dB re 1 \( \mu \text{Pa} \) in a distance of 750 m from the pile driving location. Different noise mitigation techniques are developed to decrease the hydro sound level significantly. Nevertheless, these limits are often not met yet.

All of these measures like bubble curtains or cofferdams have in common that they affect the direct transmission path between pile surface and water. During the pile driving the major part of the impact energy is transmitted into the subsoil which results in wave propagation through the soil. The retransmission of these soilborne waves into the water represents the secondary transmission path. By now, the quantitative influence of the soil parameters and the induced soil vibration on the emitted hydro sound is not evident yet.

The German research project BORA deals with the hydro sound emission and its prognosis with the help of numerical simulations. To get a better understanding of the whole physical process and all influences on hydro sound under offshore conditions, extensive offshore measurement campaigns are conducted in this project.

OFFSHORE MEASUREMENTS

The offshore wind farm BARD Offshore 1 (BO1) is located in the German North Sea, about 100 km northwest of the coast. 80 turbines of the 5 MW class were installed in a water depth of 40 m. The Tripile foundation consists of three piles with a length of 85.1 m and an outer diameter of 3.35 m, which are connected above sea level with a transition piece (Fig. 1a).

The pile installation was done with impact pile driving. For this, the hydro hammer MHU 1900S was used. Fig. 1b shows three sections of the piling process. During the first one, the impact energy was raised step by step to the maximum impact energy of 1400 kJ until a penetration depth of 19.5 m was reached. In the second section with penetration depths from 19.5 m to 31.3 m a sound mitigation technique, the small bubble curtain (SBC), was tested. The SBC system was developed by MENCK as a part of the German research project HYDROSCHALL-OFF BO1. It consists of vertical perforated tubes and one ballasted perforated ring [1]. The tubes are stored on reels and are submerged under operation. Linking the SBC system to air compressors a closed air curtain is set up around the pile, which mitigates the hydro sound level emitted through the direct transmission path. To measure the potential of noise reduction a reference measurement was done without noise mitigation technique at a penetration depth of 18.5 m (Ref1). During the testing of the SBC system at BO1 the SEL could be reduced from 179 dB (Ref1) to 166 dB at a distance of 750 m to the pile. After the testing a second reference measurement (Ref2) was done in the third piling section with lifted SBC system.

The geotechnical data showed medium and fine sands with dense to very dense bulk densities. A mixed layer of cohesive soil with parts of clay, silt and sand is located in a depth of 25.5 m to 27.5 m (Fig. 1b).
During the pile driving of one foundation pile extensive offshore measurements were done. Prior to the installation the foundation pile was instrumented onshore with a total of 48 sensors to measure the pile dynamics due to the impact excitation [1]. 35 hydrophones were placed in distances from 10 m to 1500 m to the pile in different water depths to measure the emitted propagation of the pressure wave taking the direct transmission path in the water. Additionally, Ocean-Bottom-Seismometers (OBS) were deployed with a total of 18 geophones on the sea bed to measure the soil vibration velocity induced by the secondary transmission path. Thereof 6 OBS systems with one vertical geophone in distances from 250 m to 1500 m to the pile and 4 OBS systems with triaxial geophone measurements in distances from 10 m to 66 m to the pile surface were used [1].

SOIL VIBRATION DUE TO PILE DRIVING

The triaxial geophone very close to the pile in a distance of 10.0 m (Ank1) westwards of the pile surface was linked to a hydrophone array, so the geophone data is synchronised to 13 hydrophones (Fig. 2). In the south three LOBSTER (Longterm OBS for Tsunami and Earthquake Research) systems were positioned at distances of 26.5 m (Lob1), 46.2 m (Lob2) and 65.9 m (Lob3) to the pile surface, which is considered as the sound source. The positions were detected by a multi beam sonar and correlated to the measured pressure wave propagation velocity in water of 1515 m/s during pile driving. The three LOBSTERs are synchronised to each other by a GPS signal based on the coordinated universal time (UTC). Therefore, wave propagation velocities can be calculated between the LOBSTER positions. Additionally to the triaxial geophone a hydrophone is integrated into the LOBSTERs. The orientations of the different measuring devices are calculated by plane vector rotation [2]. The sampling frequency was 44,100 Hz for the hydrophone array and 1,000 Hz for the LOBSTERs.

Fig. 3 shows the measured vibration velocities of the triaxial geophone positions for one single blow due to impact pile driving at a penetration depth of 18.5 m with decreasing amplitudes and with increasing distance to the sound source.
At position Ank1 the soil signal is superposed by the pressure wave of the direct transmission path. The hydro sound pressure induces the geophones. The surface wave of the secondary transmission path with a lower wave propagation velocity had no time to separate from the waves of the direct path due to the short distance to the sound source.

This separation effect is visible at measuring positions Lob1 to Lob3 (Fig. 3). Here three different wave initiations can be seen with different wave propagation velocities. The wave initiation I reaches position Lob1 about 250 ms prior to the impact blow and results probably from hydraulic lift of the drive block. In fact this wave front is not relevant for the emitted hydro sound noise and is disregarded in further evaluation.

The measured wave initiation II is induced by the hydro sound pressure wave. The wave propagation velocity of 1515 m/s was calculated by the measured conductivity and
temperature of the water as well as the water depth during pile driving. The sound immission of the direct transmission path excites the geophones and it is assumed that a transmission into the marine surface layer is possible [3]. Due to technical limitations the sampling rate was 1,000 Hz, which causes aliasing at such high wave propagation velocities.

Wave initiation III results from the secondary transmission path pile-soil. The horizontal vibration velocities $v_x$ of the sea bed in radial direction are higher than the vibration velocities of the induced movement by the hydro sound pressure wave. Because of the unknown pathway it is not possible to calculate the exact wave propagation velocity of wave front III at a known travel time. On the supposition of different pathways starting at the pile toe or the penetrated pile surface with possible reflection at deeper soil layers like the cohesive soil layer at a depth of 25.5 m, the wave propagation velocity can be estimated to 250 m/s. This corresponds to propagation velocities of shear waves, which are induced by the pile surface friction, or of Scholte waves, which are surface waves.

With advancing penetration depth of the pile from Ref1 to Ref2 the vibration velocity $v_x$ of wave initiation III at position Lob3 increases from 7.9 mm/s at 18.5 m up to a maximum of 17.2 mm/s at 30.0 m (Fig. 4). That means an increase of about 120% during constant impact energy [2]. The amplitude of the vibration velocity in wave II induced by the direct transmission path is reduced during the operation of the SBC system. But the noise mitigation measure has no effect on the amplitudes of wave III. Here, the penetration depth of the pile is the main influencing parameter while the impact energy is constant during the shown impact blows.

This significant amplitude increase did not occur at the closer measuring positions Lob1 and Lob2, where an undistinguished scattering of 20% is recorded. As a result of the triaxial measurements a sound radiation angle of the sea bed as well as a vibration velocity $v_{xz}$ of the wave III can be calculated by vector addition for each position Lob1 to Lob3. Figure 5 shows the amplitude increase of the vibration velocity at a distance of 65.9 m to the pile with rising penetration depth, while in distances of 26.5 m and 46.2 m the amplitude hardly changes. Due to a small vibration velocity $v_z$ in vertical direction a sound
radiation angle $\Phi$ up to $5^\circ$ can be calculated. This angle also increases significantly at Lob3. The evaluation of the results leads to the assumption of wave “tunnelling”. This means that subsoil waves induced by impact pile driving are able to come up to the sea bed a couple of meters away from the pile due to wave reflexion and refraction. The influence on the hydro sound immission can in some cases develop only at far distances dependent on the three dimensional soil layering, while measurements close to the sound source do not necessarily record the whole sound radiation.

Fig.5: Velocity amplitude $v_{xz}$ of the surface wave and sound radiation angle $\Phi$ on the sea bed dependent on the penetration depth of the pile in 18.5 m (top) and 30.0 m (bottom).

POTENTIAL SOUND RADIATION OF THE SEA BED

The propagation of primary wave fronts in water and soil results in a head wave problem which radiates underwater noise shown by Dahl and Reinhall [4]. Besides the primary wave fronts, which are generated by impact pile driving associated with the Mach cone, also shear waves and surface waves are propagating through the soil. The measured surface wave (wave initiation III) due to impact pile driving also has an effect on the hydro sound pressure.
Fig. 6 shows only the time dependent wave front III beginning at 600 ms related to the signals in Fig. 4. There is a hydro sound pressure which can be correlated directly to the soil vibration velocity $v_{xz}$. With an increasing pile penetration depth from 25.0 m up to 33.0 m the hydro sound pressure reacts to the increasing amplitude of $v_{xz}$ at Lob3.

Due to the free submerging of the measuring devices the relative orientation between the three geophones and the hydrophone is unknown. The geophones can be located as close as the hydrophone to the pile, closer or further afar. That is why the soil vibration signal starts a little bit earlier at Lob3 and later at Lob1, where the hydro sound wave front of the sea bed radiation is measured before the soil vibration.

**CONCLUSION**

Offshore measurements of soil vibration velocities during impact pile driving were done to investigate the influence of the secondary transmission path on the radiated underwater noise. The mitigation of the wave taking the direct transmission path pile-water by pile-surrounding barriers is in most cases insufficient to reach the German limitations of hydro sound immission. Therefore, it is necessary to investigate the qualitative and quantitative effect of the secondary transmission path pile-soil-water.

The triaxial measurements done with geophones positioned in distances from 10 m to 70 m to the sound source reveal upward sound radiation angles $\Phi$ up to $5^\circ$ of the surface wave front at the sea bed. Furthermore, an amplitude increase of the soil vibration velocity was recorded with increasing distance to the pile and increasing pile penetration resulting from wave reflexion and refraction at deeper soil layers.

The synchronized measurements showed a direct influence on the hydro sound pressure due to soil vibration. Surface waves like Scholte waves have an effect on the underwater noise, which is qualitative shown by the measurements.
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For further information, please visit www.bora.mub.tuhh.de.

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THE EVANESCENT PRESSURE WAVES ASSOCIATED WITH GROUND ROLL FROM SEABED IMPACTS

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\textbf{Abstract:} Disturbance of the seabed interface will radiate waves within both media. Whilst most studies of biological effects have looked at pressure waves in the water, the majority of aquatic life is more sensitive to motion, not having evolved pressure sensors.

More details will be presented on the interaction between the rolling motion of the seabed, or ground roll, and the nature of the localised pressure variations close to the seabed. These evanescent pressure changes provide a way to study this ground roll without the uncertainties due to the mounting of vector sensors such as geophones.

The water particle velocities from this mechanism will be concentrated near the seabed and can be larger than the sediment motion.

\textbf{Keywords:} Evanescent, interface waves, water particle velocity.
1. SEABED IMPACTS CREATE SLOW INTERFACE WAVES

The concerns over seabed impacts such as those due to off-shore wind turbine piling, led to studies of the vibration modes of the seabed. Measurements in 2012 of the emissions from a test pile in a dock at Kinderdijk provided data on slow waves constrained to the interface between the Rhine silt and the water.

![Three axis geophone data from the seabed 68m from a test pile.](image)

Results shown by Hazelwood & Macey [1] show the high frequency high speed bulk sound waves arriving at the geophone array. Whilst not calibrated for this, the array is sensitive to this arrival, and shows the delay before the onset of “ground roll” vibration. This is dominated by energy in a low frequency band (15-40 Hz). The rapidly decaying piling impact energy has been transformed into a long low frequency wavelet structure by the soft sediments in the dock. It should be noted that such shallow water cannot propagate pressure waves of this frequency, due to the duct cut off effects.

2. MODELLING THE SEABED AND THE GROUND ROLL

With better measurements still urgently needed, limited funds have been used for modelling. The detailed features of the Kinderdijk results reflect the complex environment of the dock walls and mooring structures discussed in 2013.

Similar motions have been modelled by using a graded sediment structure. Jensen et al [2] reproduced data on waves recorded in the Mediterranean by workers at SACLANT. They shared some features, but were of even lower frequencies (<10Hz). The seabed in our work was modelled with data from Hamilton [3] to create layers of progressively stiffer clay with depth for the finite element analysis (FEA). This shear wave speed (Cs) profile causes refraction back towards the surface.
Fig 2 is taken from Jensen et al [2], p475, showing the wave properties in a sedimentary seabed covered by a shallow (20m deep) sea. This was used for their long range modelling, but is similar to Hamilton’s data. The shear wave speed $C_s$ is seen to increase from 60m/s to 300m/s at 80m. They simulated results from long range propagation (>2km).

![Wave properties diagram](image)

**Fig.2: Data used to achieve a best fit to results by a wavenumber integration process**

The data shown on absorption, $\alpha$, was found by a best fit process, but our short range FEA assumed none. FEA provides more detail at the expense of the range achievable. Whilst significant matches are seen, the differences have prompted further study.

### 3. THE DETAILED RESPONSE OF THE FEA MODEL

The finite element analysis can accept a force pulse of differing shapes, then showing different and often more complex results. Here a 1MN pulse has been “tuned” to provide a simple result seen at simulated ranges to over 500m. The wavelet has strong similarities to the results from Kinderdijk, but is notable for delivering most energy within 0.1 seconds.

![Wavelet comparison diagram](image)

**Fig.3: Simulated motion of the seabed 64m with consequent pressures**

This choice thus provides an enlightening display showing a very short wavelet. This mode shows one response of the graded seabed as modelled. There will be other modes, but this simple form is conducive to a detailed analysis.
Fig. 4: A simulated seabed with node 700 at 64m radius from the central forcing pulse.

The display shows the sediment as blue where there is little displacement. The exaggerated deformation is colour coded for the resultant vector displacement magnitude. The 16m depth of water above is shown green when there is little acoustic pressure, but red or blue dependent on the sign of the pressure change induced by the passing ground roll wavelet. This pressure change is largely confined to the bottom, showing an evanescent form, decaying with height. The two fields are spaced (white gap) for clarity.

Fig 5: The horizontal displacement of the water differs from the adjacent seabed.

Further analysis of the same FEA model shows how the water slips past the adjacent solid. The vertical motions are identical, but the horizontal displacements and velocities of the water are much larger (~ threefold), for the rolling motion, seen after 0.6 seconds. Although smaller, the horizontal motions for an earlier arrival (before 0.5 seconds) are in opposition with the water moving out (positive), as the solid moves in (towards the source), an even more surprising result.
In practice there will be a boundary layer, in which the discontinuity of the model will be distributed with depth. Such boundary layers are usually thin, unless there is large scale turbulence. Piercy [4] showed how the thickness depends on the Reynolds number and kinematic viscosity, which for water is about $10^6 \text{ m}^2/\text{s}$. Given the short time for such a layer to build, the layer will be less than 1cm thick, so that the FEA can ignore it.

However, more detailed calculations are required to study its significance to wildlife. Many seabed creatures could have body parts in both regions when active. This slippage ("rip-slip") will then provide another alarming sensation, and probable cessation of activities such as feeding.

4. THE LOWER FREQUENCY MODES

Discrepancies between the earlier data (SACLANT) and later results have been investigated using the FEA models. Lower frequency modes are initiated by a force pulse of longer duration than is likely to be made by a piling blow. Whilst unrealistic for piling, the lower absorption may allow these modes to dominate at longer distances, as seen in the earlier work with explosions.

Fig 6 shows a 4 Hz mode which dominates for impulses lasting longer than 0.1s. This wave also travels at about 100m/s, again using a seabed layered using a Cs profile similar to that reported by Hamilton. It thus has a wavelength of about 25m. The view encompasses a 256m distance and shows depths to 40m within an axisymmetric seabed model of 512m radius and 128m depth.

![Fig.6: A lower frequency, longer wavelength mode distorts deeper sediments.](image)

5. THE RETENTION OF ENERGY WITH TIME

The refraction due to the speed profile with depth is analogous to the “surface duct” which occurs when surface waters are well mixed and isothermal. Then the increase in wave speed with pressure at depth gives rise to “cylindrical spreading” of energy, and a consequential increase in range over the spherical spreading of sound in other cases. However, duct borne sounds lose their coherence, with impulsive sound (such as
explosions) being “stretched” (Urick [5] p161) in time. In contrast the ground roll waveform largely retains its form, with minimal energy loss.

![Figure 7: After 1.5 s, the total wavelet energy becomes stable at 533 Joules](image)

This “energy coherence” is seen in waves described as “solitons”, especially gravity waves such as those formed as internal waves on a thermocline (Urick [5] p157).

An investigation of this energy coherence was made using a solid only model (HL11t4) wherein the energy contained in the wavelet was computed. To do so a “soliton box” was defined to track the wavelet as it progressed outward from the point at which the forcing impulse was applied.

![Figure 8: The total wavelet energy is calculated in a box 40m x 30m deep](image)
As expected, after the wavelet had left the origin, the two energy components, elastic and kinetic, become equal, a characteristic of a travelling wave motion, where successive volumes exchange energy between the two forms. Between 1s and 1.5s travel time, faster modes move out of the 40m x 30m deep “box” for which the energy is calculated, and total energy becomes stable at 533 joules. Fig 8 shows the situation after 3s. The analysis terminated after 4s, when the wavelet core has travelled 448m, approaching the edge of the 512m radius of the model. Some evidence of (false) reverberation occurs in Fig 7, due to reflections from the boundaries of the model.

6. THE ABSORPTION OF DIFFERENT MODES

Whilst surface ducting can lead to extended ranges for higher frequency acoustic pressure noise, the limited evidence for the ~20Hz ground roll vibration is that absorption will play a large part. Much lower frequencies such as those of earthquakes suffer less loss, and very long period waves are detectable after passing around the Earth several times [6]. They have a period of ~4 minutes, and correspond to flexure of most of the Earth’s crust, rather than a few metres at the seabed. The energy coherence of this characteristic wavelet is seen to contribute to this remarkable detection range.

A 2013 land trial using saturated London clay, attempted to mimic the subsea sediment, but the absorption was found to be very strong. Ground roll wavelets were recorded of similar frequency (20-40Hz), form and speed (~ 100m/s), but they were indetectable after travelling 40m despite the choice of a substantial deposit of this clay, a stratum over 100m deep as shown on the geological survey, extending over > 300m range (Pinks Hill near Guildford).

In contrast the wavelets seen at 68m range in Kinderdijk were easily recorded, but the site was complex and not conducive to extensions of the planned work programme. However, the techniques developed would allow such measurements to be made if suitable site permissions and funding were available.

7. CONCLUSIONS

The earlier work has been extended by further analysis showing details of the simulated ground roll waves which should be capable of being measured in reality.

The low frequency pressure oscillations near the seabed in shallow seas will compete with sound generated by surface waves, but should show a marked decay with height above the sediment with a hydrophone array. Accelerometers or geophones, also deployed as a vertical array, should show the rip-slip discontinuity in horizontal particle velocities.
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REFERENCES

EFFECTIVE REDUCTION OF OFFSHORE PILING NOISE

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Abstract: High underwater noise emissions from offshore piling are potentially harmful to marine life and can reach dangerous levels in a large area. An increasing number of erected and prospected offshore wind turbines needs effective noise reducing methods to achieve the German BSH standard level of 160 dB SEL at a distance of 750 m from pile driving.

The innovative method of hydro sound dampers (HSD) uses curtains of robust air filled elastic balloons with high underwater noise reduction effects and special PE-foam elements with high dissipative effects from material damping, to reduce impact noise. The resonance frequency of the elements, the optimum damping rate for impact noise, the distribution and the effective frequency range can be fully controlled.

The physical model of the development and radiation of underwater piling noise, the background of noise mitigation methods and the theoretical background of hydro sound dampers for the mitigation of continuous noise and transient noise is explained.

Offshore tests have already shown the high potential of HSD-noise mitigation even in the lower frequency range of today’s large hydraulic hammers between 50 and 350 Hz. In the first half year of 2014 a new HSD-noise mitigation system will be applied to the monopiles of an offshore wind farm in the North Sea. To reduce the mobilization time the complete HSD-system is hanging below the hydro hammer. It is expected that this first serial offshore application will demonstrate a new effective way to reduce offshore piling noise.

Keywords: Hydro Sound Dampers (HSD), offshore piling, underwater noise mitigation VibrationKeyword1, keyword2, keyword3, etc
1. INTRODUCTION

Underwater noise from impact piling results in considerable sound emissions and is distinguishable above ambient noise over distances of several tens of kilometers from the source as given in [1]. The construction noise of offshore wind turbines is potentially harmful to marine life, in particular to marine mammals. Different zones of underwater noise immissions can be defined in the surrounding of a source of acoustic noise. The ranges of zones depend on the hammers and possible noise mitigation methods.

Due to larger piles requiring higher driving energies, even higher underwater noise levels are expected in future offshore projects and this is also accompanied by an increasing number of erected offshore wind turbines. Effective noise reducing methods are in great demand, getting sound levels below recommended acoustic emission thresholds that are no longer harmful and disturbing to marine mammals and other protected animals.

2. UNDERWATER PILING NOISE

Numerical simulations and measurements of underwater noise during the construction of offshore foundations such as monopiles, tripods, tripiles and jackets are used to study the generation, radiation and attenuation of underwater noise.

Most of the piling energy is driven into the sea ground. Only about 1%-2% of the whole ram energy is radiated directly from the wet surface of the pile into the surrounding water inducing very high underwater sound levels. Depending on the properties of the ground material one part of the impact energy is radiated indirectly from the sea ground into the water, resulting in additional underwater sound. Numerical simulations and measurements of the impacts wave in the pile after Figure 1 show that the resulting travelling wave is reflected up to several times at stepped cross sections of the pile and at both ends of the pile until all the kinetic energy is damped out and is radiated into the ground. The radiated underwater noise is propagating into all directions of the shallow water, reflected at the free water surface and at the sea ground after [1].

![Fig.1: Travelling impact wave within the pile inducing UW-sound waves and reflections at the free surface of the water and at the sea ground after [1].](image-url)
The underwater piling noise is usually described by the peak Sound Pressure Level (peak SPL) in decibels (dB) of the maximum instantaneous positive or negative sound pressure $|p_{\text{peak}}|$ of the measured impact noise that is referred to the underwater sound pressure of $p_0 = 1 \mu \text{Pa}$

$$\text{peak SPL} = 20 \log \left( \frac{|p_{\text{peak}}|}{p_0} \right) \text{ in dB re:1} \mu \text{Pa}$$

and by the Sound Exposure Level SEL in decibels (e.g. dB re:1$\mu$Pa$^2$s), which is an equivalent energy level of the noise of a single pile driving impulse, based on $T_0 = 1$s.

$$SEL = 10 \log \left( \frac{1}{T_0} \int_0^{T_2} \left( \frac{p(t)^2}{p_0^2} \right) dt \right) \text{ in dB re:1} \mu \text{Pa}^2\text{s}$$

Measurements of the underwater piling noise show peak levels of more than 210 dB (SPL) re 1$\mu$Pa and sound exposure levels of more than 180 dB (SEL) re 1$\mu$Pa$^2$s at a distance of 750 m from pile driving sites, depending on ram energy and pile size.

3. UNDERWATER NOISE MITIGATION

Several underwater noise mitigation methods are developed in the last years. Nearly all of them use the underwater noise mitigation effects of air bubble curtains. Small air bubbles in the water are released from perforated tubes with compressed air at the bottom of the seabed. Surrounding the piles these air bubble curtains can reduce the radiated noise using the effects of acoustic impedance mismatching of the bulk acoustic impedance change and of frequency dependent resonance scattering effects of the bubbles.

The air bubble curtain attenuation of high frequency noise above 1 kHz is very high. But the broadband sound level of the piling noise mainly depends on the lower frequency noise, far below 1 kHz, where the attenuation from bubble curtains is only poor.

The reasons for this are, that large air bubbles (several cm) with low resonant frequencies are uncontrollable, showing chaotic movements and dividing themselves when they are slowly arising to the surface of the water.

Against this, only constant shapes of air bubbles and nearly steady state resonant excitations are able to achieve high sound attenuations of resonant air bubbles. As a consequence there is no benefit from high theoretical underwater sound attenuation potential of large air bubbles with low resonant frequencies, when using conventional air bubble curtains. That is why the attenuation of high frequency noise is very high above 1 kHz, but only poor in the most important lower frequency range of hydraulic hammers with the highest 1/3-octave SEL-levels between 100Hz - 300Hz.

4. HYDRO SOUND DAMPERS (HSD)

To overcome these problems, a new underwater noise reducing method is developed, using gas filled envelope bodies and PE-foam elements as hydro sound dampers, instead of free natural air bubbles as described in [2]. The size of the bodies, the effective frequency range, the damping rate, the number and distribution of the hydro sound
dampers (HSD) and the influence from hydrostatic pressure can be fully controlled, if the envelope bodies are fixed to a pile surrounding fishing net or to stiff frames.

The efficacy of HSD in reducing underwater noise depends on the frequency and the volume rate of the hydro sound dampers. Rates of about 1-2% of the HSD are sufficient to obtain good results. At these volume rates vertical forces from buoyancy and horizontal forces from tide currents are still small.

In contrast to free air bubbles, hydro sound dampers as gas filled bladders and PE-foam elements allow to use three different physical reasons for effective underwater noise attenuation:

- Reflections of sound waves at impedance steps from water, filled with HSD-elements,
- Resonance effects with high scattering, multiple reflections and absorption of sound,
- Dissipation of acoustic waves according to material damping effects of HSD elements.

The important resonant effect with high scattering, multiple reflections and effective absorption of sound waves in the water is to be seen in Figure 2. The very strong interaction of a vibrating HSD-element and the surrounding water is to be seen at the water surface in Figure 2. This interaction also takes place under water as shown in Figure 3, but it is not visible there.

Hydro sound dampers are used in the whole frequency range of pile driving noise from 50Hz - 5000Hz. It is possible to control the damping rate, the size, the number and the distribution of the HSD around the pile. Finally, HSD-systems don’t need compressed air supply.

5. CONTINUOUS NOISE ATTENUATION OF BUBBLES AND HSD

The theory of bubble acoustics is used to develop a new effective noise mitigation method as air is the best medium to reduce underwater sound.

The resonance frequency $f_{Res}$ of an air bubble in water at depth $z$ meters and of the bubble radius $a$ is depending on the ambient water density $\rho_A = 1.03 \times 10^3$ kg/m$^3$, the ambient static pressure at the depth $z$ of $p_A = 10^5 (1+0.1z)$ N/m$^2$ and the ratio of specific heats $\gamma = c_p/c_v = 1.4$ for the enclosed gas under constant pressure $c_p$ and volume $c_v$ is approximated by the following Minnaert equation:

$$
f_{Res} = \frac{1}{2\pi a} \left( \frac{3\gamma p_A}{\rho_A^2} \right)^{1/2}.
$$

(3)
For an air filled HSD-element with the additional stiffness $S$ of the membrane or of the PE-foam material the resonance frequency is:

$$f_{\text{res}} = \frac{1}{2\pi a} \left( \frac{3\gamma}{\rho_s} + S \right)^{1/2}.$$  \hspace{1cm} (4)

The whole extinguished power $\Pi_e$, as the sum of scattered sound power $\Pi_s$ and absorbed sound power $\Pi_a$ of a bubble or of an air filled HSD-element is referenced to the incident plane wave intensity $I_{\text{inc}}$ of the bubble to get the extinction cross section $\sigma_e$ after [5]:

$$\sigma_e = \frac{\Pi_e}{I_{\text{inc}}},$$  \hspace{1cm} (5)

With the damping constant $\delta$ as the sum of all damping constants due to re-radiation or scattering $\delta_r = k a$ plus thermal conductivity $\delta_t$ and shear viscosity damping $\delta_v$ the extinction cross section $\sigma_e$ can be calculated after [5] from:

$$\sigma_e = \frac{\Pi_e}{p_{\text{inc}}^2/\rho_s c} = \frac{4\pi a^2 (\delta/\delta_r)}{\left( f_{\text{res}}/f_{\text{inc}} \right)^2 - 1 + \delta^2},$$  \hspace{1cm} (6)

where the extinguished power is obtained from the rate at which work is done on the element by the incident pressure $p_{\text{inc}}$. At resonance the extinction cross section $\sigma_e$ of a bubbly element is much larger than its physical geometric cross section. Taking into account of additional material damping $\delta_m$ that is usually much larger than all other damping terms, the resonance peak of the extinction cross section function $\sigma_e$ is reduced very much and the attenuation effect is mainly due to material damping.

The extinguished power of an incident plane wave of intensity $I_{\text{inc}}$ by each bubble or element is $I_{\text{inc}} \sigma_e$. Assuming $N$ resonant bubbly elements of the same size, the change of intensity over a distance $dx$ is:

$$dI = -I_{\text{inc}} \sigma_e N \, dx$$  \hspace{1cm} (7)

The integration of the change of intensity over $x$ results in:

$$I_x = I_{\text{inc}} e^{(-\sigma_e N x)}$$  \hspace{1cm} (8)

and for the sound pressure in:

$$p_x^2 = p_{\text{inc}}^2 e^{(-\sigma_e N x)}$$  \hspace{1cm} (9)

After [5] the change in sound pressure level $\Delta \text{SPL}$ of a sound wave is:

$$\Delta \text{SPL} = 20 \log_{10} \frac{P(x)}{P_0} = -10 \sigma_e N x \log_{10} e$$  \hspace{1cm} (10)

after traversing the distance $x$ in water due to bubbles or elements of the same size where $\sigma_e N$ is the extinction cross section of the bubbles per unit volume of the water. The attenuation per unit distance due to $N$ bubbly elements per unit volume is:

$$\alpha_b = -\frac{\Delta \text{SPL}}{x} = 4.34 \sigma_e N \text{ dB/distance.}$$  \hspace{1cm} (11)
Figure 4 shows examples of attenuations in dB/m of single air filled membranes per unit volume with about 15% material damping and of different sizes. Therefore, also the resonance frequencies and the volume rates of air are different.

The attenuation curves in Figure 4 refer to continuous parts of offshore piling noise. Numerical examples of impact noise and transient waves from offshore piling and examples of measurement results of offshore applications are explained in [4] and [6].

**REFERENCES**


Abstract: Environmental Impact Assessment (EIA) of ocean energy projects is not only a legal requirement but also an assurance of the projects sustainability, a promoter of public acceptance and a benefit for industry, making the project more attractive to investors and governments who traditionally have seen environmental concerns as a barrier. Noise is one of the environmental descriptors of high relevance and concern in any project or anthropogenic activity and ocean energy projects are not an exception. The increasing number of wave farms or demonstration areas becomes an opportunity to learn, analyze and evaluate potential environmental impacts including the underwater noise impact on the marine environment. In general, a best practice to assess the underwater noise of a given source involves its characterization, the characterization of the baseline acoustic environment, the acoustic transmission path and the characterization of the receptor. Due to the underwater noise variability it is necessary to analyse long data sets of measurements which might not be available or might not be possible to acquire in a short period of time. As ocean energy is still in a demonstration phase, the project’s deployment period is usually uncertain and sometimes for a short period, making it difficult to coordinate monitoring programmes in order to accomplish authorities’ requirements in terms of EIA process deadlines. This study aims to review three monitoring plans that were or are being applied to the underwater noise assessment of the wave energy projects in Portugal and Spain in order to develop a methodology that fits the permitting requirements without compromise acoustic data quality.

Keywords: wave energy devices, underwater noise assessment, acoustic monitoring plan
1. INTRODUCTION

Since 1940s efforts have been made in research and development of technologies to harness wave energy. However, only recently different types of devices are being tested in real environment [1] and beyond its technological viability also environmental uncertainties need to be addressed [2].

Most marine animals use underwater sound to perceive and interact with the environment and thus the increasing of underwater noise levels might have an adverse effect on them [3, 4, 5]. Currently, there is a lack of knowledge regarding the impact of the underwater noise produced by wave energy devices, at temporal and spatial scales, in the marine environment.

The impacts of human activities in the environment are established in the EU legislation through the Environmental Impact Assessment (EIA) Directive (Directive 2011/92/EU of 13 December 2011). This was developed to identify potential environmental impacts of the projects and implement measures to reduce or mitigate them. Although the EIA Directive does not address specifically wave energy projects, its implementation in some European countries has been adapted to cover these projects. This has contributed to the sector promotion and sustainability making it more attractive to investors and governments who traditionally saw environmental concerns and licensing as a barrier.

Nowadays, underwater noise monitoring programmes are designed to include baseline measurements, the acoustic characterization of the source and its propagation through the marine environment. Although guidelines are suggested for underwater noise measurements of offshore wind technologies [6] there are no standard methodologies for wave energy devices.

One of the limitations of underwater noise monitoring of wave energy devices is related with the uncertainty of the project’s deployment duration since sometimes is difficult to accomplish the monitoring plan within the short periods of time that demonstration projects are usually installed. Another constraint is related with oceanographic and meteorological variability that largely contributes to underwater noise levels and the propagation conditions and should be part of the analysis.

This paper reviews the methodologies used to describe acoustic baseline conditions and underwater noise of three wave energy devices, while discussing their suitability for the impacts assessment. The methodologies used for the acoustic baseline assessment for two case studies will be addressed: the WaveRoller oscillating wave surge converter located in Peniche, Portugal and the OPT point absorber power buoy (WavePort project) to be installed in the BIMEP test site in the North of Spain. Another case study for the evaluation of underwater noise monitoring methodologies involves the acoustic campaign of the oscillating water column in Azores, Portugal.

2. USED METHODOLOGIES FOR BASELINE MEASUREMENTS AND SOURCE CHARACTERIZATION: CASE STUDIES

2.1. Underwater acoustic baseline measurements

The WaveRoller (WR) is a device that converts ocean waves to energy and electricity through the back and forth movement of the composite panel when water is driven by wave
surge process. The machine operates in near-shore areas (approximately 0.3-2 km from the shore) at depths of between 8 to 20 meters. It is fully submerged and anchored to the seabed. As the WR panel moves and absorbs the energy from ocean waves, the hydraulic piston pumps attached to the panel pump the hydraulic fluids inside a closed hydraulic circuit [7].

Two devices, at a time, have been installed in the west coast of Portugal at Peniche and an acoustic monitoring plan have been established for the last device installed, including baseline measurements (acoustic background), device noise measurements and propagation.

In September 2010, baseline measurements were carried out in the predicted deployment location and at multiple positions nearby in order to characterise the acoustic conditions of the place. Five hydrophones were fixed to the seabed at mid-water depth along two transects of 3 km extension. Also, deployments from a boat were carried out along one of the transects.

Measurements to characterize WR noise emissions should have been carried out along the same transect with several fixed hydrophones, as well acoustic transmissions to assess the transmission loss. Unfortunately, the device has been installed during a short period of time and it was not possible to carry out measurements to characterize the emitted noise.

The OPT PowerBuoy is a wave energy point absorber device based on modular, buoy-like structures capable of responding to different wave conditions. Each PowerBuoy is a semi-submerged floating buoy, consisting of a spar, surrounded by a moving toroidal float, moored with a three point anchor system and connected to shore via a submarine power cable. The buoy is designed to produce its rated output in a 4 metre significant wave height environment, and is suited to operate in water depths of 50 to 70 metres [8].

The OPT PowerBuoy baseline measurements were carried out, in June 2012, along two transects centred on the future position of the device. Two hydrophones at two depths were used and operated from the boat. To characterize the noise source measurements will be carried at a fixed position near the device, using two autonomous hydrophones deployed in a fixed position. Also, two transects, centred in the device and an extension of a maximum of 2 km will be carried out.

2.2. Underwater noise source characterization

Underwater noise measurements to characterize the noise emitted by a WED device were carried out for the Oscillating Water Column converter, the Pico Plant, installed onshore at the Pico Island (Azores, Portugal). The plant consists of a hollow reinforced concrete structure – pneumatic chamber – above the water free surface that communicates with the sea and the incident waves by a submerged opening in its front wall, and with the atmosphere by a fibre duct with an air turbine. The incident waves cause vertical oscillation of the water column inside the chamber, which in turn causes alternate air flow to and from the atmosphere, driving the turbine and the generator attached to it [9].

The measurements were carried out on May 2010, using a fixed autonomous hydrophone installed at 10 m distance from the device and at 9 m depth. The period of measurements included intervals of device operation and intervals when the device was off. In September 2010, a second campaign was performed to assess the device’s noise propagation. With that purpose two transects were carried out, with a maximum extension of 3200 m, and the hydrophone was operated from a boat. The closest distance to the source was 200 m. The noise emitted by the device was not detected at any distance although a slight increase in the SPL was detected at the closest distance. However, these results have been identified as being more related with the bathymetry and the sea state during measurements than with the noise emitted by the device.
3. RECOMMENDATIONS FOR UNDERWATER ACOUSTIC AND NOISE MEASUREMENTS

The information about the noise emitted by WEDs is scarce and it is important to take advantage of the demonstration projects to fill this gap. Besides, the environmental conditions, the project characteristics and the type and number of available equipment (hydrophones) to carry out the measurements might limit the monitoring activities. The following recommendations are based on the existence of two autonomous hydrophones.

3.1. Baseline measurements

Underwater noise baseline measurements are important to know noise fluctuations that are likely to occur due to natural and anthropogenic (mainly distant ship noise) sounds variability at the site when the device is not installed. Currently, standards for underwater ambient noise measurements are still being developed by the International Organization for Standardization (ISO) but there is a general agreement about the parameters that should be considered in ambient noise measurements [10]. Nevertheless, it should be kept in mind the objective of underwater ambient noise measurements for acoustic monitoring plans of WEDs. Baseline measurements can be carried out through drifting, fixed sampling stations or by a combination of both.

Fixed positions are preferable if measurements need to be taken in harsh conditions. In this case, it is recommended to deploy one hydrophone near the site where the device will be installed and other at a fixed distance where it is not predicted to have the influence of the noise of the device if it is installed [6].

If the device is already installed, background ambient noise measurements might be carried out near the device when it is off or at a distance from which the noise of the device cannot be detected. The distance will be dependent of the site conditions but it is recommended to take into account the sediment composition, bathymetry, and spatial distribution of vessels which should be similar to those find in the device site before its installation. However, as a matter of consistency it is preferable to perform baseline measurements before the device is installed.

The use of drifting measurements is recommended when there is a significant bathymetry variation since sound propagation depends on depth. Also, drifting measurements along transects are useful to systematically record ambient noise and use measurements to validate acoustic models.

The combination of fixed and drifting measurements might be useful to get information of ambient noise near the site where the device will be installed and at different distances from it. The main limitation is related with good sea conditions that are needed to operate in a safety mode. The selection between these three experimental configurations should also account on the environmental conditions at the site.

3.2. Source measurements and propagation

WEDs’ noise propagation is strongly dependent on the sea state conditions since a more energetic sea state leads to an increase in background noise levels. The longer the period of deployment, the greater is the likelihood to record the noise emitted by the device in different
sea state conditions. Therefore, the experimental configuration for data acquisition should have this in consideration in order to characterize the noise in different operating conditions.

During harsh sea state conditions data acquisition through fixed autonomous hydrophones is more suitable to characterize underwater noise of WEDs. According to the device design they can fall into two physical categories: bottom/surface or water column distributed. Depending on it, two approaches are possible: two distances or two depths [10]. Two distances configuration fits well when the device is installed on the seabed or onshore, as the OWC and the WR. Two depths fits well when the device is in the water column, as the OPT PowerBuoy. If two depths are used it is recommended to deploy one hydrophone at ¾ and other at ½ of the total depth [6]. If one depth is used and the device is near the seabed the hydrophone might be deployed at ¾ of the total depth. If the device is on the surface the hydrophone should be positioned mid-water depth. This allows reducing the bottom and air/sea-surface interactions.

The autonomous hydrophones location should be selected taking into account a balance between the increasing of background noise levels and the detection range of the emitted noise. The closest the distance to the device the great the likelihood to record the sound emitted by the device in a more energetic sea state. However, safety considerations as well as far field and near field issues also need to be in account.

Considering the uncertainty regarding the device deployment during a period long enough to allow the noise measurements to be carried out and the cost effectiveness of the monitoring activities, the noise characterisation of the source can be coordinated with the noise propagation measurements. In this case, one hydrophone can be deployed at mid-water depth near the device and other used to carry out transects from the device. It is worth to notice that if transects are going to be carried out, sea state should ensure data acquisition in safety conditions.

Transects have usually an extension of 3 km and measurements can be used to validate propagation models. However, the distance between the sampling points within the first 300 m from the device should be smaller than the distance between sampling points until the end of the transect. This would allow a better characterisation of the device noise propagation near the device, especially during more energetic sea state conditions.

Finally, it is important to consider the distribution of maritime traffic in the vicinity of the project. The sites where the devices are installed usually have a surrounding restricted area to maritime traffic and this might represent an increase in sound pressure levels with distance from the device as the sampling points approach the limit of such area. Bathymetry slope and seabed type should also be considered in such analysis.

4. LESSONS LEARNED

As ocean energy is still in a demonstration phase, the technologies’ installation date and deployment period are usually uncertain. Sometimes the devices are installed for short periods of time (until problems occur and they have to be removed from the site to be repaired) making it difficult to coordinate monitoring programmes in order to accomplish the authorities’ requirements in the context of the EIA process. Nevertheless, all possible measurements are useful to improve methodologies to measure the underwater noise emitted by WEDs.

It is expected that single wave energy devices (WEDs) could introduce low frequency sounds in the environment but not at levels that might impact marine species. In the case of the Pico OWC plant, although higher sound pressure levels have been detected when the device was operating it was not possible to identify the device noise within more than few
hundred meters from it (200 m). The background noise resulting from the sea state conditions may in some cases mask the noise emitted by the wave energy devices (WEDs). Since the objective is to carefully characterise the noise produced by WEDs, it is necessary to take into account the sea state conditions when determining the right distances (sampling points) from the device at which the noise can be identified in order to conclude about its propagation.

Depending on the campaign objectives, transects can be carried out using fixed sampling points or drifting and either continuous or duty cycle recordings. However, a duty cycle configuration is more suitable for fixed measurements and continuous recording for drifting measurements. If a duty cycle is defined during drifting transects there is a risk to miss a cycle between sampling points bringing a delay in the campaign schedule or being the acoustic recording contaminated by the research vessel’s noise. Therefore, continuous recordings can be a good solution in drifting measurements. It is worth to note that the hydrophone should be carefully synchronised with the current time to facilitate the records data selection and analysis based on devices deployment timing.

If the main objective is to characterize the noise emitted by the device it is reasonable to use a fixed sampling point near the device for a period of time that allows to characterize it in different conditions. If more than two hydrophones are available it is possible to deploy them at different distances allowing measurements at different distances at the same time contributing to assess the noise propagation.

Although lacking international standards for underwater noise measurements, there are some relevant international standards and agreements that considered the impact of underwater noise in the marine environment. Some examples are the ISO standards (under development), the Marine Strategy Framework Directive, the OSPAR Convention and ICES:09 report, and recent guidelines of the International Maritime Organization [11, 12, 13, 14].

This paper is an attempt to address the issues related with methodologies to measure the underwater noise emitted by WEDs and that might be used for the impacts evaluation process when underwater noise is being an increasing issue of concern.

5. ACKNOWLEDGEMENTS

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UNDERWATER SOUND LEVELS AT A WAVE ENERGY DEVICE TESTING FACILITY IN FALMOUTH BAY, UK

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Abstract: There is a paucity of evidence on the noise produced from in situ wave energy converters (WECs) during all stages of their deployment, operation and decommissioning. Research in this area is needed to inform the consenting process. The aim of this research is to gather empirical data to address this knowledge gap. A WEC has been trialled at the Falmouth Bay Test Site (FaBTest) in Cornwall, UK since March 2012. The area supports considerable commercial shipping and recreational boating along with diverse marine fauna, including bottlenose dolphins, harbour porpoises and fish. A passive acoustic monitoring device, recording broadband sound in the effective frequency range 10 Hz to 32 or 48 kHz, for half an hour in every hour, has been deployed at the FaBTest site since March 2012. Underwater sound monitoring covered a two week baseline period, a five day installation period, testing periods when the WEC was producing power and when the device was not producing power but was in situ. The median received sound levels during the baseline period ranged from 52-71 dB re 1 µPa in the frequency range 0.01-10 kHz, then decreased to \~32.23 dB re 1 µPa at 48 kHz. It is likely that the considerable shipping present at the site affects the sound levels. Received levels were, on average, higher during installation activity compared to periods of no installation activity with a median increase of 8.2 dB re 1 µPa (interquartile range = 6.7 dB re 1 µPa) in the frequency range 10-5000 Hz. Average sound levels were found to be louder at times when the WEC was producing power compared to times when the device was in situ and not producing power in the frequency range 10-4000 Hz with a prominent peak in the frequency range 57-63 Hz. From the long term monitoring of the site it has been identified that the sound levels are highly variable, and it is difficult to determine the effect of the wave energy converter in such a variable ambient noise environment.
Keywords: marine renewable energy, wave energy, underwater sound

1. INTRODUCTION

Marine renewable energy is a relatively young industry. The majority of research on the subject of marine renewable energy and underwater sound to date has occurred in the offshore wind industry. The wave energy industry is at a much earlier stage of development with few commercial deployments although it has potential to expand [1].

There is a paucity of evidence to inform environmental impact assessments (EIA) regarding sounds originating from wave energy converters (WECs) during all stages of device deployment, operation and decommissioning [2].

Underwater sound levels have been monitored at the Falmouth Bay testing facility (FaBTest), a site designed for the testing of marine renewable energy devices in a moderate wave climate. Sound levels were recorded during a two week baseline period prior to the deployment of the WEC, a five day installation period, and periods of device operation and non-operation in situ. The WEC monitored is a point absorber with three power take off units (PTOs) or power generation units, which are each moored to the seabed.

2. METHOD

2.1 Location

The passive acoustic monitoring (PAM) device was deployed at the Falmouth Bay test site (FaBTest) site on Cornwall’s south coast, UK. The site is 2 km\(^2\) in size, between 3 and 5 km offshore and 20 to 50 m in depth.

Falmouth Harbour is a busy commercial port with 1,193 ship arrivals reported in 2012 [3] which is the second highest in the South West, UK. The area also supports considerable recreational boating [4] and diverse marine fauna including bottlenose dolphins, harbour porpoises, seals and fish [5].

2.2 Data gathering equipment

Two Autonomous Multichannel Acoustic Recorders (AMAR Generation 2; Jasco Applied Sciences) were deployed at FaBTest, alternately, at a distance of ~200 m from a wave energy converter. There were seven deployments, of which six are reported on here due to a loss of the data from the sixth deployment. These devices use GeoSpectrum M8E hydrophones which were calibrated by Jasco Applied Sciences prior to deployment. The AMAR was programmed to record for the first 30 minutes in every hour. Table 1 shows the deployment locations and equipment settings. The effective frequency range is 10 Hz to half the sampling frequency. There were two methods of deployment; the dome configuration and flotation collar configuration. The resolution of the recording is 24-bit.

For the dome method, the AMAR was attached to a custom built weighted steel frame which rested on the seabed. For the flotation collar method, a flotation collar was attached around the AMAR which causes it to float in vertical position in the water column. The AMAR was attached to the centre of a weighted ground rope and was approximately 5 m off the seabed. During deployments three to seven the hydrophone cage was covered with a cloth
hat to reduce flow noise. As there were different deployment configurations which could have affected the received levels (RLs), for example, the hydrophones were at different depths between the dome deployments and flotation collar deployments, the datasets were kept separate for analysis.

<table>
<thead>
<tr>
<th>Deployment number</th>
<th>Date of Deployment</th>
<th>Position (Degrees and Decimal Minutes)</th>
<th>Sampling frequency (kHz)</th>
<th>Deployment method</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>10th March 2012</td>
<td>N50.099720 W04.996390</td>
<td>96</td>
<td>Dome</td>
</tr>
<tr>
<td>2</td>
<td>13th June 2012</td>
<td>N50.098889 W04.995278</td>
<td>64</td>
<td>Flotation collar</td>
</tr>
<tr>
<td>3</td>
<td>20th August 2012</td>
<td>N50.100409 W04.996118</td>
<td>64</td>
<td>Flotation collar</td>
</tr>
<tr>
<td>4</td>
<td>8th November 2012</td>
<td>N50.100633 W04.995900</td>
<td>64</td>
<td>Flotation collar</td>
</tr>
<tr>
<td>5</td>
<td>09th January 2013</td>
<td>N50.101256 W04.996308</td>
<td>64</td>
<td>Flotation collar</td>
</tr>
<tr>
<td>7</td>
<td>04th June 2013</td>
<td>N50.100283 W04.997333</td>
<td>64</td>
<td>Flotation collar</td>
</tr>
</tbody>
</table>

Table 1. Deployment date, locations and equipment settings of the AMAR

2.3 Data processing

Following retrieval of the AMAR, the data were downloaded from the device to a computer and converted to .wav files using proprietary software.

The acoustic data were calibrated using the hydrophone response curves provided from the manufacturer’s calibration and interpolated to provide a calibration value per Hz, with an acoustic gain of 0 dB.

MATLAB scripts were developed to process the .wav files. These include a fast Fourier transform (FFT) function, using a 1 s Hann window with a 50% overlap, performed for every file to provide the power spectral density (PSD) in dB re 1 µPa² Hz⁻¹ s⁻¹ and square pressure (pRMS²). The mean and median PSD value in dB and the mean of the square pressure is calculated per minute per Hz and stored. The sound levels per second are also stored for periods of interest. The median was then found per Hz for every minute. The median is used here as the data is not normally distributed.

2.4 Site Activity Periods

The baseline period occurred immediately prior to the deployment of the WEC from 11th – 25th March 2012. Installation activity is considered to include all activities associated with the installation of the device and included the presence of work vessels on site, the laying of anchor chain and other activities. This activity took place intermittently over the five days 26th-30th March 2012. Comparison periods, when no installation activity was taking place, were chosen based on similarity in time and wave height in order to minimise differences in sound levels due to other factors.
Operational activity is considered to occur when one or more PTO systems are active and producing power as recorded by the device developer. This occurred intermittently between periods of non-operational activity.

### 2.5 Ancillary data

Tidal data is obtained from the POLPRED depth averaged High Resolution UKCS Model (CS20-15HC) for location 50° 6' 32.2" N  4° 59' 28.6" W  (National Oceanography Centre).

### 3. RESULTS

#### 3.1 Baseline

![Baseline sound levels 11\textsuperscript{th}-25\textsuperscript{th} March 2012. The sound levels are the mean square pressure sound level per half hour sound file and the tide speed is the POLPRED modelled tide speed for the site.](image)

The median sound level during the baseline period ranged from 52.17-71.18 dB re 1 µPa Hz\textsuperscript{-1} in the frequency range 0.01-10 kHz, then decreasing to ~32.23 dB re 1 µPa Hz\textsuperscript{-1} at 48 kHz. The half hourly mean square pressure sound levels range from 34.01 dB to 113.42 dB re 1 µPa (Fig. 1).

#### 3.2 Installation

Sound levels were, on average, higher during installation activity compared to periods of no installation activity with a median increase of 8.2 dB re 1 µPa (interquartile range = 6.7 dB re 1 µPa; Fig. 2) in the frequency range 10-5000 Hz. The maximum minute median sound levels during installation activity were 123.67 dB re 1 µPa at 245 Hz which coincided with the installation vessel on site preparing to deploy one of the anchors nearest the AMAR. During this half hour file, the maximum received level from square pressure per second was 129.23 dB re 1 µPa\textsuperscript{2} Hz\textsuperscript{-1} s\textsuperscript{-1} at 323 Hz.
3.3 Operational

Received levels are louder, on average, during operational periods compared to non-operational periods in the frequency range 10 Hz to ~4 kHz. There is a prominent peak at 62 Hz, which has a median difference of > 6 dB. The median received level at 62 Hz is 65.10 dB re 1 µPa during operational activity. It is quieter, on average, during operational periods compared to non-operational periods above 4 kHz.

4. DISCUSSION

During baseline monitoring, sound events of > 100 dB were identified which are likely to originate from local shipping or industrial activity (Fig. 1). Peaks in the sound levels < 100 Hz correspond to peaks in tidal speed (Fig. 1) indicating that flow noise affected the sound levels during periods of high tidal flow during the first deployment. A cloth hat was used to reduce the flow noise in deployments 3 to 7.
As converting to dB levels from square pressure preserves the order, the median level is equivalent when calculated over dB levels or square pressure levels [6]. The median received levels were consistently louder at all frequencies during periods of installation activity compared to similar periods of no installation activity (Fig. 2). Received levels of > 120 dB re 1 µPa Hz\(^{-1}\) s\(^{-1}\) were recorded during installation activity. However, installation activity was found to only occur in 23 half hour periods over five working days representing a limited period of time affected, although installation may have a greater impact if multiple devices are installed.

It is challenging to assess the impact from the operational activity of the WEC in Falmouth Bay, in which there are already varying sources of anthropogenic sound, including local shipping and industrial activity from the local port (Fig. 1). However, when the entire time period is assessed, the sound levels during operation were found to be louder than in non-operational periods on average (Fig. 3) in the frequency range 10 Hz to ~4 kHz. The prominent peak at 62 Hz, which is louder by over 6 dB on average during operational periods (Fig. 3), suggests that the sound levels at this frequency in particular are affected by operational activity of the WEC. This frequency is below that of best hearing sensitivity for high- and mid-frequency cetaceans and pinnipeds [7].

There are more half hour files of non-operational activity compared to operational activity. This may have affected the results and explain why the sound levels are quieter during non-operational activity at frequencies greater than 4 kHz. For example, testing of the WEC may not have occurred during extreme weather events.

The sound levels have been found to be variable at the site and affected by local shipping. Installation has contributed to loud sound levels and operational activity may affect the local sound levels at certain frequencies.

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CABLED OBSERVATORY ENABLED ACOUSTIC MONITORING OF HYDROTHERMAL DISCHARGE

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Abstract: New technological developments have recently enabled long-term monitoring through cabled seafloor observatories. In order to increase understanding of the complex interactions of earthquakes, volcanic eruptions, ocean waves and the hydrothermal systems, the Cabled Observatory Vent Imaging System, known as COVIS, has been designed for long-term monitoring via remote sensing of multiple types of hydrothermal discharge. The imaging and Doppler modes use backscatter from the plume itself to sense the boundaries of the plume and the vertical flow rates within the plume. The diffuse mode uses the decorrelation effect of turbulence in front of a solid object to detect areas of outflow on the seafloor. COVIS is currently installed at the NEPTUNE observatory’s Endeavour site; COVIS images the plumes rising from Grotto Vent on the Endeavour Segment of the Juan de Fuca Ridge off the western coast of North America. COVIS’s observations of plume bending and volume flux highlight the potential for long-term monitoring via remote sensing in this Marine Protected Area. Recent studies have led to the inversion of volume flux observations for heat flux in the discharging plumes and to the recognition of the transient effects of atmospheric storms on plume behavior.

Keywords: acoustic imaging, ocean observatories, scattering and fluctuations, volume scattering, inversion methods in underwater acoustics
1. INTRODUCTION

Seafloor hydrothermal systems form a fundamental part of magmatic systems, wherein seawater percolates downward into the deep crust, absorbs volcanic heat, rises buoyantly, and discharges at the seafloor [1]. High temperature fluids exit in a focused manner through sulfide chimneys forming black-smoke like plumes and through flanges on the sides of larger sulfide mounds. Lower temperature fluids, usually cooled by mixing with cold seawater just below the seafloor, exit diffusely through the sulfide mounds themselves and through faults, fissures and fractures in the seafloor.

On a global scale, such hydrothermal discharge is important for several reasons. First, the convective flow of the mid-oceanic ridge hydrothermal system is responsible for ~33% of the transfer of heat from the Earth’s crust to the ocean [1]. Second, the impact of hydrothermal discharge on ocean chemistry is on the scale of riverine impacts for many chemical processes [2]. Third, the heat and chemical output of hydrothermal discharge supports a local ecosystem through chemosynthesis based on H₂S carried by the hydrothermal fluids. Despite this, the existing estimates of heat and chemical fluxes in hydrothermal systems are mostly based on spot, small volume, or asynchronous area measurements taken as snapshots at long (1yr) intervals in a few selected sites [3].

Acoustic imaging (and related Doppler processing) has demonstrated the potential to overcome many of the limitations of earlier studies [4, 5, 6]. Mounting a multi-beam sonar on a platform capable of at least one direction of rotation enables fully three-dimensional imaging and reduces the asynchronicity of resulting data set [7].

![Fig. 1: (a) COVIS in place on the seafloor. (b) Close-up of sonar head. (c) The image combines plume imaging as isosurfaces and diffuse flow mapping as decorrelation values draped on the bathymetry. COVIS's location is given by an orange triangle.](image)

Long-term acoustic monitoring requires an ongoing power supply, a method of data retrieval to keep local hard-drives from overfilling, and a stable platform with a rotation system. New technological developments such as underwater junction boxes, ROV (Remotely Operated Vehicle) underwater mateable fiber/copper connectors, and new fiber optics protocols have recently enabled long-term monitoring through cabled seafloor observatories such as The North East Pacific Time-series Underwater Networked Experiment (NEPTUNE) observatory [8]. The NEPTUNE observatory supplies power (~10 kW), high data-bandwidths (~10 Gbs), and interactive access through standard internet protocols along its fiber optic cables. Its installation at the Endeavour Segment of the Juan de Fuca Ridge via Ocean Network Canada (http://www.oceannetworks.ca) enables scientists to study the response of a hydrothermal system to geological events.
through continuous long-term, high temporal resolution observations using both acoustic and traditional sensors.

Here we describe a new sonar platform for remote sensing of hydrothermal discharge, review the acoustic methods, and present some early monitoring results (Figure 1).

2. METHODOLOGY

The Cabled Observatory Vent Imaging System (COVIS) is designed to 1) visualize the behavior of the high temperature focused plume, 2) quantify the outflow fluxes of the same, and 3) map low temperature diffuse hydrothermal discharge near the seafloor. A multi-beam, high resolution sonar is used to make these measurements acoustically.

2.1. COVIS platform design

COVIS was specifically designed to connect to the NEPTUNE Canada cabled ocean observatory which provides 375VDC and 1 Gigabit Ethernet interface to the instrument through a submarine cable. COVIS is a 4-meter tall titanium tripod on which the sonar and cable observatory interface systems are mounted. A unique 3-degree of freedom (pitch, roll, yaw) mechanical rotation system allows for the sonar transducers to be aimed appropriately at the hydrothermal vent and mechanically scan the sonar across the vent plume. The rotation system allows a large range of pitch, roll, and yaw angles for observing any structures within the sonar range at the installation site.

The projector produces a 400Khz, 1 degree (vertical width) by 128° (horizontal width) fan beam with 256 receive beams of width 0.5° and spacing 0.5° when making measurements of the plume extent and during collection of data for Doppler processing for plume velocities. The sonar boresight is mechanically stepped in 1° increments to image the plume from the vent orifice to an upward angle of 60°. A different projector is used with a vertically much broader beam to make measurements of the diffuse flow around the base of the vent structures. This projector produces a 200 kHz, 30° (vertical width) by 128° (horizontal width) fan beam with 128 receive beams of 1° width and 1° spacing.

Metadata from rotation motor position sensors and a sonar-mounted attitude sensor (magnetic heading, gravity sensing pitch/roll) along with system health information is streamed during operation for real-time monitoring. Sonar data are stored locally on the platform with a modest capacity (several days). NC Data Management and Archiving (DMAS) software periodically uploads the sonar data files making best use of network bandwidth and decoupling the real time sonar data acquisition. Sonar data are post-processed upon delivery to shore automatically with several plume image and Doppler data products available via the NC data management website (http://oceannetworks.ca).

2.2. Acoustic methods

COVIS scans the effluents from a hydrothermal system every three hours in three different modes (i.e. Imaging, Doppler and Diffuse flow). At each degree step within an Imaging or Doppler scanning cycle, the transmitter sends pulsed signals towards the focused plume of interest and receives the backscatter from the plume particulates and turbulence-induced acoustic-impedance fluctuations.

The intensity of the backscatter signals are calibrated for the volume scattering strength (VSS) in units dB•m⁻¹. The analysis of the Doppler frequency shift observed in
the backscatter signals using a covariance method yields the plume velocity component along the acoustic line-of-sight ($V_r$) [5]. The $V_{SS}$ and $V_r$ obtained at successive degree steps are stacked together and interpolated to a uniform 3-D rectangular grid with 0.25 m (Imaging mode) or 0.5 m (Doppler mode) intervals in all three dimensions. Rendering the isosurfaces of the gridded $V_{SS}$ data generates 3-D plume images.

A geometric conversion of the gridded $V_r$ data gives the vertical velocity component ($W$) of the plume [5, 9]. Further analysis of $V_{SS}$ and $W$ yields key plume physical properties, including plume radius ($b$), volume flux ($Q$), and expansion rate ($Ex$) [7, 9]. Subsequently, the initial buoyancy flux ($B$) of the plume can be estimated based on the classic formula for a buoyancy driven plume [9]. Knowing $B$, the heat flux ($H$) driving the plume can be calculated as

$$H = C_p \rho B/g/\alpha_T,$$

where $C_p$ is the specific heat capacity, $\rho$ is the reference seawater density, $g$ is the gravitational acceleration, and $\alpha_T$ is the thermal expansion coefficient.

In the Diffuse flow mode, the 200 kHz transmitting/receiving pair scans the seafloor in a near horizontal direction. The Acoustic Scintillation Thermography (AST) method [10] is applied to detect and map the areal distribution of diffuse hydrothermal discharge near the seafloor from the decorrelation between the backscatter of consecutive sonar scans.

3. INSTALLATION

In September 2010, COVIS was deployed at 47°57′N, 129°6′W, approximately 30 m to the northeast of the Grotto mound, a hydrothermal vent cluster on the Endeavour Segment of the Juan de Fuca Ridge in the Northeast Pacific and connected to the NEPTUNE observatory. COVIS sits between jumbled blocks of basalt looking up at the Grotto and Dante vent sites. This ability to minimize interference in the vent ecosystem while still quantitatively monitoring the hydrothermal discharge is an asset in the Endeavour Marine Protected Area.

The Grotto mound is a large hydrothermal sulfide structure (area ~450 m$^2$) hosting intense hydrothermal venting. Near the western rift valley wall, 4-5 high temperature focused vents and ubiquitous diffuse discharge are observed on top of a 10 m tall edifice named the North Tower. The effluents from the focused vents and part of the diffuse discharge merge into an integrated plume, which is the primary target of COVIS and is called the North Tower plume hereinafter. Near the eastern end of Grotto, there is weaker plume, which is the secondary target of COVIS.

4. SCIENTIFIC RESULTS

An extended and ongoing sequence of 2D and 3D acoustic data has been collected by COVIS including 26 days after its installation in 2010 and a continuing time series since September 28, 2011. The extraction of plume radius and bending from the 3D volume of scattering strength verifies that these hydrothermal plumes expand vertically and respond to ambient currents more or less as predicted by simple plume models [4, 7].

A skeleton-based centerline algorithm estimates the path of the plume core(s), the overall plume orientation, and the plume expansion [11]. Plume expansion ($Ex$) is a measure of the vertically averaged entrainment rate and varies significantly with the speed of ambient currents [6]. Tidal currents also account for much of the variability in the orientation of the plume centerline. However, several interruptions to this regular
variability have been observed. Figure 2 shows two cases where flow appears to be consistent for two days, which were detected in an automatic analysis [12]. Possible explanations include currents generated by the passage of atmosphere storms or the appearance of the background influx to the rift valley during low tidal amplitudes.

![Figure 2: The 2010 season time series of plume bending is plotted as north-south (N) and east-west (E) components. Plume N component shows strong regularity (likely tidal) interrupted by occasional days of constant direction flow (marked by red stars).](image)

Figure 2: The 2010 season time series of plume bending is plotted as north-south (N) and east-west (E) components. Plume N component shows strong regularity (likely tidal) interrupted by occasional days of constant direction flow (marked by red stars).

Figure 3 (a) shows a 28-month (Oct 2011 to Jan 2014) time series of the volume flux of the North Tower plume at varying levels above the base of COVIS. The overall mean volume flux increases from 3 m$^3$/s to 6 m$^3$/s over the 8 m rise shown in Figure 3 (a) due to the entrainment of ambient seawater into the plume during its buoyant rise. Figure 3 (b) shows considerable short-term variations (standard deviation ~36% of the mean) in the time series of the heat flux driving the North Tower plume. The heat flux has no apparent trend (mean ~ 18 MW).

![Figure 3: (a) 28-month (Oct 2011 to Jan 2014) time series of the volume flux (Q) of the North Tower plume at varying level, Z, above the base of COVIS. (b) 28-month time series of the heat flux (H) driving the North Tower plume.](image)

Fig. 3: (a) 28-month (Oct 2011 to Jan 2014) time series of the volume flux (Q) of the North Tower plume at varying level, Z, above the base of COVIS. (b) 28-month time series of the heat flux (H) driving the North Tower plume.

5. CONCLUSIONS AND FUTURE PLANS

COVIS's observations of plume bending and volume flux highlight the potential for long-term monitoring via remote sensing in this Marine Protected Area. Recent studies have led to the inversion of volume flux observations for heat flux in the discharging plumes, which provides a key constraint for modelling magmatic-hydrothermal processes beneath the vent field. Efforts to identify geologic events have led to the recognition of the transient effects of atmospheric storms on plume behavior. Further work will investigate potential correlations between plume variability and local earthquakes.
6. ACKNOWLEDGEMENTS

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Session 8

Advances in Acoustic Measurement Systems: Technologies and Applications

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THE SMO ANTENNA: STATUS AND FIRST RESULTS

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Abstract: Submarine Multidisciplinary Observatory (SMO), deployed on March 2013 at about 100 km off Sicilian coasts, is a cabled underwater acoustic antenna composed of 10 large bandwidth (10 Hz – 70 kHz) hydrophones, installed at different depths, from about 3150 m to about 3350 m. The goal of SMO (https://web.infn.it/smo) is a long term monitoring of the underwater acoustic environment for studies on astrophysical, biological, geophysical fields. Acoustic signals acquired from sensors are continuously sampled off-shore and transmitted through optical link to the shore station for real-time analysis and recording. Thanks to an innovative data acquisition system based on underwater time stamping, all acoustic sensors are synchronized and phased with GPS time. First results and future extension towards a very large acoustic array (hundreds of sensors) will be presented.

Keywords: neutrino, telescope, acoustics, hydrophone, GPS

1. INTRODUCTION

Within the activities of the INFN-NEMO Collaboration [1] and of the KM3NeT Consortium [2] a demonstrator for a deep-sea astrophysical neutrino detector has been deployed on March 2014, at 3500 m water depth, offshore Capo Passero (Sicily). The demonstrator, called NEMO Phase-II, has been design to measure the visible Cerenkov photons originated by charged particles propagating at velocities greater that the speed of light through the seawater by means of a an array of photomultipliers (PMTs). The charged particle track is reconstructed measuring the time of arrival of the Cherenkov photons on the PMTs of known position. PMTs are installed on board a mechanical structure (tower), consisting of a sequence of horizontal frames (named floors), made of marine grade aluminium, interlinked by a system of ropes, anchored on the seabed and kept vertical by appropriate buoyancy on the top (see Fig.1). The vertical distance between two adjacent floors is 40 m; a spacing of about 100 m is added between the lowermost floor and the base of the detector [3]. The monitoring of the mechanical structures, moving
under the action of the underwater currents is provided by an acoustic positioning system, composed of 4 acoustic beacons anchored on the sea-floor in known positions, and an array of 10 broad-band (10 Hz ÷ 70 kHz) acoustic sensors installed along the detector at different depths, from 3150 m to 3350 m. This array, called Submarine Multidisciplinary Observatory (SMO) [4], has been designed to allow, in addition to the positioning of the underwater structures, the monitoring of the underwater acoustic environment for studies on astrophysical, biological, geophysical fields. In particular, the activities of the SMO Collaboration are addressed to the studies on acoustic neutrino detection, the passive acoustic monitoring of cetaceans and the development of novel tsunami detection techniques [5].

![Mechanical scheme of the NEMO – SMO detector.](image)

2. THE SMO ACOUSTIC ARRAY

The SMO acoustic array consists of 10 SMID TR-401 custom piezo-ceramics hydrophones, developed for the SMO and NEMO Collaborations by the SMID Company. The 10 hydrophones of the SMO array are part of a set of 40 hydrophones and preamplifiers that have been acquired, tested and certified at the NATO- Centre for Maritime Research and Experimentation (CMRE). Their receiving sensitivity at 1 bar pressure has been measured between 5 kHz and 70 kHz, coupling the hydrophones with a SMID AM-401(V)1 preamplifier, having gain of +38 dB. As shown in Fig.2, the overall sensitivity of the hydrophone-preamplifier system is almost flat in the whole frequency range and it is -172±3 dB re 1 V/μPa. The relative change of sensitivity as a function of pressure, is less than ± 1 dB up to 400 bar (4000 m water depth equivalent), negligible for SMO purposes [6]. The data acquisition system of the SMO acoustic array is fully integrated with the NEMO Phase II detector one. The analysis of the acoustic signals is entirely performed on-shore. All
acoustic signals, acquired by the acoustic sensors, are sampled underwater and continuously sent to shore through the NEMO Phase-II data transmission system. A schematic view of the data acquisition system for a single floor of the NEMO-SMO detector is shown in Fig. 3. In each floor of the detector are installed two acoustic sensors. For each acquisition channel, signals from the acoustic sensor and its respective preamplifier are digitized underwater digitized at 192 kHz rate with a resolution of 24 bit by a professional audio Multi-bit Delta-Sigma Crystal CS-5381. The output signals from the analogue-to-digital converter are encoded into the AES/EBU standard stereo protocol by a Digital Interface Transmitter (DIT) Crystal CS-8406. The use of a standard protocol makes it easy to manage the audio data with professional sound boards and commercial software and firmware reducing the costs and implementations time with respect to custom protocols. The digital AES/EBU data stream is continuously sent to the Floor Control Module Board (FCMB). The FCMB labels the AES/EBU audio blocks (containing 192 audio samples) with the GPS time received from shore, embedding it in the audio protocol and sent them, together with data from optical modules, to shore through a point to point optical link. In particular, the GPS time is written in the 192 bit control word composed by the user bits of a single AES/EBU block. The resolution of the acoustic data time labelling is 25 ns. The audio data, labelled with the GPS time of acquisition, are sent to shore in the main data stream together with the optical modules and oceanographic instruments data through the whole data transmission system of NEMO Phase-II detector. The data stream for each acoustic sensor is about 6.3 Mbit/s [8].

Fig. 2: Sensitivity curves for 40 SMID TR-401 hydrophones as a function of frequency (at 1 bar pressure measured) at NATO-CMRE [7].

Fig. 3: The NEMO-SMO data acquisition chain. The data acquired by the underwater detector is continuously transmitted to shore via optical link for real-time analysis.
3. SMO TIME SYNCHRONIZATION

One of the tasks of SMO array is to provide through acoustic triangulation the positions of the NEMO Phase II structures, detecting the signals emitted by a long baseline (LBL) of acoustic beacons anchored on the sea-floor. Positioning performances are related to the accuracy on the measurements of Time of Arrival (ToA) of the beacon signals at the hydrophones. It depends on the accuracy of the latency time of the whole data acquisition system. The latency time of whole data acquisition system is due to: 1) the electronic boards’ latency, 2) the hydrophone’s ceramics latency and 3) the time delay along the data transmission system. Latency time of SMO data acquisition electronics has been measured for each acquisition channels, sending test electrical waveforms to the input of the electronic front-end at known time by means of a waveform generator triggered by the Pulse Per Second (PPS) signals of the same GPS receiver used for the data time-stamping. This latency time is given by the difference between the time written by the acquisition system in the data stream in correspondence of the arrival of the test waveform and the time of wave emission, derived by the PPS signal. A latency time of 39.5±0.1μs has been measured for all channels of SMO array. Measurements exhibit no variation as function of input signal amplitude and frequency. The latency introduced by mechanical movements of the hydrophones ceramics has been evaluated through tests at the water-pool facility at the CNR-IDASC using a calibrated emitting source. The cumulative ceramics response time of the hydrophones and of the emitting ceramics used in the test is about 25 μs, equivalent to a distance of ~ 4 cm, calculated taking into account the sound velocity in the water. The time delay due to the data transmission system for each single acquisition channel has been evaluated with a precision of about 100 ps, knowing the length of the cables and the optical paths of the system [9].

4. MEASUREMENTS IN DEEP SEA

SMO is working in deep sea environment since March 2013. The continuous stream of acoustic data from each hydrophone is analysed in real time for positioning purposes. In addition 5 minutes per hour of unbiased raw data from all sensors are stored on a digital library for off-line analysis. In order to evaluate the acoustic noise in the deep sea, all the recordings are studied trough spectral analysis. The spectrogram of each file is calculated, dividing the recordings into slices of 216 samples (34 ms), using an overlap of 50% and Hanning windowing. For each slice the periodogram is calculated using a Fast Fourier Transform at 65546 points. Fig. 4 reports a spectrogram of a 5 minutes long recording from the hydrophones F6H0, installed at a depth of about 3150 m. This peculiar recording shows beacon pulses and sperm whale sounds. Unfortunately the SMO data acquisition system is also sensitive to the electromagnetic interferences induced by the underwater instrumentation, that produce horizontal lines in the spectrogram visualisation. This issue has been solved for the future activities of the SMO Collaboration.

Measurements in sea water have confirmed the nominal sensitivities of the SMID TR-401 hydrophones (Fig. 5). The sensitivity of each sensor has been calculated through the analysis of the acoustic signals from external beacons (5 ms long sinusoidal pulses at 32 kHz), placed at about 400 m far from tower base. The measurement has been performed taking into account the geometrical attenuation and the absorption related to the ionic relaxation of MgSO₄ and B(OH)₃ dissolved in sea water.
Fig. 4: Spectrogram of a 5-min long recording from hydrophone F6H0 (about 3150 m water depth). In this recording beacon pulses (32 kHz) and sperm whale clicks are easily detectable.

Fig 4: Sensitivity of SMO acoustic sensors calculated from the positioning acoustic pulses of the two external acoustic beacons of the NEMO – Phase II detector, placed at about 400 m far from tower base (red and green dots). The grey boxes refer to the nominal sensitivity of the sensors provided by NATO-CMRE.
5. CONCLUSIONS AND PERSPECTIVES

Tests and first results from deep sea show that SMO acoustic antenna has optimal performances in terms of frequency and time resolution. Therefore, it enables rigorous analysis of impulsive and continuous acoustic signals in a very large frequency bandwidth. Recordings from more than one year of activity are under analysis, this analysis will allow for the first time the long term monitoring of the underwater acoustic noise at depths greater than 3000 m. Moreover, the development, within the SMO project, of a new semi-automatic algorithm to identify and extract in real time sperm whale sounds from raw data time will support ecological studies on sperm whales population in a strategic area of the Mediterranean Sea.

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REFERENCES

PROCESSING STRATEGIES FOR EVALUATING THE SHIP RADIATED NOISE USING AN UNDERWATER VERTICAL ARRAY

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\textbf{Abstract:} In the present paper a post processing software tool developed in Matlab\textsuperscript{®} is presented to elaborate the outputs of a portable system made by a vertical array of hydrophones aimed at measuring the underwater noise emissions of vessels. The inputs of the processing software are the GPS signals of both the supply and the target vessel, and the volt signals from the hydrophones. The results after the software processing not only cover the needs of the ANSI/ASA S12.64-2009/Part1 standard but it is also able to provide a wider range of information on the acoustic emissions of the vessel such as both vertical and horizontal directivity. Finally the possibility of using the array for a spatial filtering technique, in order to improve the SNR, is presented.

\textbf{Keywords:} underwater radiated noise, ship noise, vertical array, post processing
1. INTRODUCTION

The reduction of all types of vessel emissions has become an important issue during the past decade. More recently the term “emissions” has been extended to include underwater noise because of its potential impact on marine fauna [1] pushing the shipping sector to face with such a problem. Surface vessels radiate underwater noise mainly due to machinery on board, propulsion mechanism and hydrodynamic flow around the ship hull and appendages [2]. The EU has established the Marine Strategy Framework Directive, MSFD, to investigate and implement programmes of measures which are designed to achieve or maintain ‘Good Environmental Status’ in the marine environment. The ASA S12 Committee on Noise Standards Working Group (WG) 47 has been organized to develop a commercial standard for the measurement of underwater noise from ships. In 2009 the S12 Committee submitted a draft document [3] of procedures for the measurement of underwater sound pressure levels from a surface ship at a prescribed operating condition.

The International Maritime Organization (IMO) with its Marine Environment Protection Committee (MEPC) recognized the need of an International Standard for the underwater radiated noise measurement from merchant ships and both the Technical Committee ISO/TC 43 [4] and the Technical Committee ISO/TC 8 are working on developing an International Standard to respond to that demand.

An array of hydrophones deployed by a supply vessel represents a flexible solution to carry out measurements of underwater noise in compliance with many of the recommendations and procedures addressed within the regulatory framework mentioned above. An example of such a system was extensively presented in [5].

A post processing software tool developed in Matlab® to elaborate the outputs of the hydrophones array is presented in the present work. The tool covers both the needs of [3] and provides a wider spectrum of information on the acoustic emissions of the vessel such as both vertical and horizontal directivity. Finally the feasibility and possible advantages from a spatial filtering technique are illustrated.

2. NOMENCLATURE

\[ F_s = \text{sampling frequency [Hz]} \]
\[ v = \text{ship speed [m/s]} \]
\[ z_{1,2,3} = \text{hydrophones depth} \]
\[ h_{\text{dist}} = \text{horizontal distance between the ship and the hydrophones array} \]
\[ \phi_{1,2,3} = \text{vertical angle among ship and hydrophones} \]
\[ \theta = \text{horizontal angle between ship and hydrophones array [see Fig.1 (right)]} \]
\[ \psi = \text{target vessel heading [°]} \]
\[ d_{\text{cop}} = \text{horizontal distance between ship and hydrophone array at the closest point of approach} = \min(h_{\text{dist}}) \]
\[ DWL = 2 \cdot d_{\text{cop}} \cdot \tan(30') \]
\[ DWP = \frac{DWL}{v} \text{ time interval (T}_{\text{start}}, T_{\text{end}}) \]
\[ w_{\text{hann}} = \text{Hanning window} \]
3. MEASUREMENT SYSTEM AND PROCEDURE

The measurement system consists of a vertical array of digital hydrophones with reconfigurable geometry in terms of total length, number of hydrophones or hydrophones spacing. Its design is addressed to perform measurements of underwater noise radiated from ships of different size, e.g. according to [3]. The vertical array is deployed at sea from a support vessel and recovered by it. It consists of:

- 3 digital broadband hydrophones, each of which includes a depth sensor and an electronic calibrator; the presence of the depth sensor is particularly important in order to monitor the actual position of the hydrophones during measurements reducing the uncertainties linked with the reciprocal position between target vessel and hydrophones (see e.g. [6]).
- an underwater cable of 335 m, which can be divided into 3 sections and equipped with 6 fixed connection points for hydrophones in order to easily configure the array geometry before deployment, e.g. in accordance to the length of the target ship and/or to the current water depth;
- 2 GPS antennas placed on the support vessel and on the target vessel respectively;
- a multi-channel data receiver.

The set-up scheme of the system showing all the cable sections is reported in Fig. 1 (left).

In the measurement procedure described in [3] the vessel under test (‘target vessel’) transits a straight line course with the buoy at the side both in port and starboard aspect between the COMEX (starting point) and the FINEX (ending point) to achieve the Closest Point of Approach (CPA). In Fig. 1 (right) the sketch of the test course is reported.

The time at which data start to be recorded is important for the characterization of the source directivity in the horizontal plane. As a matter of fact the time duration of records [usually performed symmetrically around the closest point of approach (CPA)] influences the range of variation of the angle formed between the buoy-ship line and the course of the ship. The longer the time record is, the wider is the range of angles, thus including data in a larger number of different reciprocal positions between the hull and the hydrophones during the ship advance.
4. SIGNALS ELABORATION

The signals processing software covers the needs of several standard procedures, providing at the same time a large number of information to make the user more confident with the results of the performed run.

Ship parameters

The outputs of the acquisition system, therefore the inputs of the post processing analysis, are the GPS signals of both the target vessel and the supply vessel, the signal from the 3 hydrophones and the sensors depth. The GPS signals are given as NMEA sentences then translated in Matlab® format. From the GPS signals relevant quantities can be derived in order to monitor the test and the behaviour of the target vessel during the measurements. Two important parameters that are to be checked after the test are represented by the ship route and speed.

\[
\theta(t) = \tan^{-1}\left(\frac{\text{Lat}_{\text{buoy}} - \text{Lat}_{\text{boat}}}{\text{Long}_{\text{buoy}} - \text{Long}_{\text{boat}}}\right) - \tan^{-1}\left(\frac{\sin(\psi)}{\cos(\psi)}\right)
\]

(1)

On the other hand the vertical relative angle is different for each hydrophone (see Fig. 3 left):

\[
\phi_{1,2,3}(t) = \tan^{-1}\left(\frac{z_{1,2,3}(t)}{h_{\text{dist}}(t)}\right)
\]

(2)
Spectra computation

The signal is digitized by the three hydrophones and stored on the pc by the acquisition system in volts. In order to obtain the power spectral density and the 1/3 octaves spectrum the recorded signal is processed as follows.

The first step consists in obtaining a time signal in Pa:

\[ S_{1,2,3}(\text{sample} \#) = \frac{S_{s1,2,3}}{10^{\text{gain}/20}} \cdot \frac{1}{10^{\text{sens}/20}} \text{[Pa]} \]  

(3)

The signal is then cut in the time interval corresponding to an horizontal angle between -30° and +30° if the analysis have to comply with [3] otherwise other angles can be set.

\[ S_{\pm 30}(\text{DWP}) = S_{1,2,3}(\frac{T_{\text{start}}}{F_s} : \frac{T_{\text{end}}}{F_s}) \text{[Pa]} \]  

(4)

Such an interval depends on the ship speed and on the horizontal distance between the ship and hydrophones array. For this segment of signal the spectrum is evaluated following the Welch procedure as below.

The signal is divided in intervals with length \( dt \geq 1 \text{ s} \) in order to obtain a spectrum of resolution of at least 1 Hz:

\[ s_i = S_{\pm 30}(dt \cdot F_s \cdot (i-1) + 1 : dt \cdot F_s \cdot i), \quad i=2:n_i \text{ with } n_i = \frac{\text{DWP}}{dt} \]  

(5)

Each interval is then multiplied by the Hanning window in the time domain and the FFT of each segment is computed:

\[ s_{\text{ht}}(dt) = s_i \cdot w_H, \quad i=2:n_i \]  

(6)

\[ Y_{\text{ht}}(1: \frac{F_s}{2}) = \text{FFT}(s_{\text{ht}}), \quad i=2:n_i \]  

(7)

The two-sided spectrum for each segment is then computed:

\[ P_i(1:F_s) = \frac{|Y_{\text{ht}}|^2}{n_i \cdot F_s \cdot \sum_{k=1}^{db} w_H(k)^2} \]  

(8)

Finally the average one sided spectrum is obtained:

\[ P(1:F_s) = \frac{2}{n_i} \sum_{i=1}^{n_i} P_i \]  

(9)
In addition the 1/3 octaves spectrum is also computed:

\[ L_k = 10 \log_{10} \left( \int_{f_{\text{ref}}}^{f_{\text{ref}}} P(f) df / p_{\text{ref}}^2 \right) \]

with \( k \) = number of 1/3 octaves bands requested (10)

The levels in Eq. 10 are the received levels and they are to be corrected for the transmission losses in order to refer the level to the reference distance of 1 metre from the source.

Transmission losses computation

In the main available standards it is usually suggested to use a simplified transmission loss law valid at each frequency such as the classical spherical law: \( 20 \log(\text{distance}) \). The aim of the present analysis is to evaluate the transmission loss in a more sophisticated way using a wave-number integration algorithm for frequencies below 1 kHz and a ray tracing algorithm for frequencies higher than 1 kHz. Regarding the environmental parameters, the celerity profile can be directly computed as the array is equipped with sensors to measure the water characteristics. As regards the bottom, in fact, the important parameters (sediment thickness and composition) are not directly monitored. Therefore the sediment thickness is derived by [7] giving information on a grid of points on the ocean bottoms with a resolution of one minute. The composition of the sediment is derived from the Deck41 database (www.ngdc.noaa.gov/mgg/geology/deck41.html), where ten different kinds of sediments are present. An example of the obtained results is shown in Fig. 4.

![Fig. 4: TL computations in the water column for the 1 kHz frequency (left); TL for the different frequencies for each hydrophone for a specific configuration (right) ](image)

In Fig. 4 (left) the values of TL are represented by the colour scale and are plotted in function of the range (horizontal distance) and depth (vertical distance) from the source. As it is clear from the picture this is a 2D evaluation of the TL and a cylindrical symmetry is considered. The value of TL for each position and each frequency can be obtained by

\[ TL = TL(f[HHz], d[m], z[m]) \] (11)

For a specific configuration of the hydrophone array the curves in Fig. 4 (left) can be computed and the received levels can be corrected to obtain the source levels as follows:

\[ L_{k \mid @1m} = L_k + TL(f_k, \bar{d}, \bar{z}) \] (12)
**Directivity**

The horizontal directivity of the ship noise emissions is obtained taking advantage of the fact that the relative angle between the ship and the hydrophones array ($\theta$, see Figures 1 (right) and 3 (right)) is changing with time during the measurement test covering a wide ranges of angles if the measurement window is sufficiently long (longer than the DWL prescribed in the ANSI/ASA standard). To compute the directivity it is necessary to iterate the procedure described in equations 1-8 to obtain the 1/3 octave levels for each second (a 1 second interval is here chosen because the GPS data are given each second). It is therefore obtained a noise level at each second and each relative angle between the ship and the hydrophone array:

$$\text{Dir}_{\text{h}@1m}(f_k, \theta) = L_k(\theta(t)) + TL(f_k, d(t), z_{1,2,3}(t)) \quad [\text{dB}]$$

$$\text{Dir}_{\text{v}@1m}(f_k, \phi) = L_k(\phi(t)) + TL(f_k, d(t), z_{1,2,3}(t)) \quad [\text{dB}]$$

(13)

The results for the horizontal and vertical directivity are reported in the polar charts of Fig. 5. As regards Fig. 5 (left) the $0^\circ$ represents the bow while $180^\circ$ represents the stern. Regarding Fig. 5 (right) $90^\circ$ is right under the keel of the ship while $0^\circ$-$180^\circ$ represent the water line. It is clear that with the measurement procedure adopted with a vertical array it is impossible to obtain levels at both $0^\circ$ and $180^\circ$.

![Fig. 5: Horizontal directivity (left) and vertical directivity (right)](image)

**Spatial filtering technique**

Data-independent beamforming is a spatial processing method that can be used to exploit the vertical array composed by three hydrophones to improve the signal-to-noise ratio. This is a crucial achievement in the frequency band from 10 to 50 Hz. Two options are considered here: a delay-and-sum beamforming with uniform weight coefficients, steered toward the ship hull; a filter-and-sum beamforming with superdirective performance, steered toward the ship hull. In the latter case, the use of the optimum weight coefficients (i.e., the coefficients that at each frequency value maximize the directivity under the constraint that the white-noise gain is higher than 0 dB) has been assumed. The white-noise gain constraint assures robustness against the hydrophone positioning errors. Fig. 6 shows the achievable directivity versus the frequency. The directivity represents the improvement of the signal-to-noise ratio when isotropic noise and plane wave propagation are assumed.
5. CONCLUSIONS

In the paper a post processing strategy for the underwater noise from ships measured with a vertical array is presented. The software developed in Matlab® covers not only the needs of [3] but gives a set of extra data regarding both the target ship behaviour during the measurements and more sophisticated data of the underwater noise emissions characteristics. Quantities regarding the ship trajectory, route and speed are fundamental to check the quality of the test and its fulfilment of the norms requirements. The additional information regarding the noise emission, such as the horizontal and vertical directivity, give a more detailed picture of the characteristics of the source emission giving the possibility of detecting the position of the most important sources on board the ship. More over the flexibility and accuracy of the system is highlighted by the possibility of performing a spatial filtering that can significantly increase the signal-to-noise ratio in the frequency band where the propeller contribution can be usually found.

REFERENCES

Development of ultra wideband dolphin speaker

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Abstract: The bandwidth of most conventional transducer using echo sounder is narrowband which relative bandwidth (bandwidth/center frequency) is about 0.1 values, less than one octave bandwidth. The recent demand of broadband transducers has been increased in the field of underwater acoustic detection to improving signal to noise ratio and range resolution. Broadband transducers have also been required in the dolphin vocalization research works and/or in the high resolution echo sounding research works. Until now scientists studying dolphin vocalizations have used low frequency (10kHz~50kHz) and narrowband (<20 kHz) underwater speakers, which can reproduce only a part of vocalizations dolphins use (over several tens Hz to 150 kHz), for playback experiments to dolphins. If a broadband transducer could be developed, it would enable to reproduce a variety of dolphin vocalizations and enhance understanding of them. We had developed the transducer covering 20 kHz – 150 kHz in 2011 for echo sounder. However this transducer do not have enough bandwidth for our purpose of playback of various dolphin vocalizations. Therefore we have applied the creative technique and tried to design ultra wideband transducer that has four octaves bandwidth. The design and characteristics of 4-octave transducer were described. And also the application of ultra wideband transducer is introduced.

Keywords: broadband, echo sounder, transducer, dolphin, playback,
1. Introduction

Recent keyword of underwater acoustic technologies is broadband. The conventional echo sounder uses monotone pulsed signal applied to the narrow band transducer. It has long history and some progress. Recent demand in underwater acoustic application of transducer would be broadband. The reason why broadband is desired might be generated from scientist side or produced by engineer side. Many of scientists felt insufficient to narrowband system and they require broadband system that might be produced new discovery. Otherwise many of engineers propose the broadband system to scientists as users that produces much information of frequency region. These proposal and requirement are synthesized and broadband technology and application would be progressed to the future. We developed the transducer that has four octaves bandwidth from 8 kHz to 150 kHz. The creative method is applied to this transducer called “ultra wideband speaker”. It is designed based on Langevin theory well known.

2. Principle of broadband

The element that resonates in single frequency was used for the transducer that transmits the ultrasonic wave used for the conventional fish finder. For example, the element that resonates to 200 kHz is carrying out form like Fig.1 using the lead zirconate barium piezoelectric element. Moreover, the transducer used for a 50 kHz fish finder is called Langevin type transducer as shown in Fig.2, makes sandwich construction an element with the resonance characteristic in high frequency with metal, and it is using resonance frequency for low-pass, lowering it.

As the frequency characteristic of this transducer is shown in Fig.3 and Fig.4, sensitivity other than resonance frequency is low with a sharp resonance characteristic. Although the Langevin type transducer was made into sandwich construction and has lowered resonance frequency, naturally it resonates also near the resonance frequency that an element has. Fig.5 expresses the frequency characteristic of transmitting sensitivity when the Langevin...
type transducer that has resonance in 50 kHz using an oscillating element with the resonance frequency of 200 kHz is designed. It is a calculation result when Q factor of the resonance characteristic that an element has is set to 10 values in the example and Q factor of the Langevin type transducer is also set to 10 values. Although a vertical axis is a relative value of transmitting sensitivity, compared with the sensitivity of an element simple substance, Langevin type sensitivity usually worsens. Thus, in Q=10, two resonance characteristics appear and it cannot produce broadband of the whole transmitting sensitivity. Fig.6 is an equivalent circuit of the Langevin transducer. It has resonance frequency $f_{r_1}$ of an element simple substance, and resonance frequency $f_{r_2}$ as a Langevin transducer, and it is thought that transmitting sensitivity is proportional to the equivalent resistance $R_1$ and the electric power consumed by $R_2$. Therefore, transmitting sensitivity $S_{TX}$ of this Langevin type transducer is shown by formula (1).

$$S_{TX} = \eta_1 P_{TX1} + \eta_2 P_{TX2}$$  \hspace{1cm} (1)

Here, $\eta_1, \eta_2$ are efficiency of electric to acoustic conversion, $P_{TX1}, P_{TX2}$ are shown by equation (2) by transmitting power sensitivity.

$$P_{TXi} = v_{TXi}^2 / |Z_i|$$  \hspace{1cm} (2)

$v_{TX}$ is transmitting voltage, $Z_i$ is impedance of equivalent circuit

$$Z_i = R_i + j(\omega L_i - 1/\omega C_i),$$  \hspace{1cm} (3)

$$|Z_i| = \sqrt{R_i^2 + (\omega L_i - 1/\omega C_i)^2}$$  \hspace{1cm} (4)
Q factor $Q_l$ is shown in

$$Q_l = 1/\omega C_d R_l.$$  \hspace{1cm} (5)

$\omega$ is angular frequency and $C_d$ is damping capacitance. The resonance frequency of an element itself and Langevin type resonance frequency were brought close, and when each Q factor was small, calculation confirmed that sensitivity in the meantime did not fall so much, either. Although the resonance frequency of $Q=5$ and the Langevin type in 120 kHz is the frequency characteristic of the sensitivity in $Q=2$ in 30 kHz, it can be said that the fall of the sensitivity between both resonance peaks remains in about 10 dB, and can realize broadband characteristic. From this, it has checked broadband transducer by theoretical calculation for it to be able to realize by forming the Langevin type transducer using the low element of Q factor.

3. Prototype of ultra wideband transducer

Based on the principle drawn for foregoing paragraph, broadband transducer made as an experiment. The layered piezoelectric device used for an actuator used for the transducer element used for the production. A layered piezoelectric device an element developed in recent years layers piezoelectric material of lead zirconate barium and obtained big displacement magnitude. Compared with the piezoelectric device of the conventional monolayer, measurement shows that Q factor of this element is low. Since a metal-electrode layer mixes between lead zirconate barium materials in order to laminate, the reason Q factor is low is considered because Young's modulus as the whole element becomes low compared with the thing of a monolayer material. Next, in order to make this layered piezoelectric device into the Langevin type structure and to make the resonance characteristic on the low frequency side, the test piece was made and the resonance length of the Langevin type transducer and

Fig.7 Double resonant characteristic of prototype transducer

Fig.8 Test piece

Fig.9 Characteristic of transmitting power sensitivity
the relation of resonance frequency were investigated. Fig.8 is a construction drawing of a test piece. The 10mmx10mmx10mm layered piezoelectric device was put into the acrylics pipe of 24 mm of outside diameters, and order was made sandwiches with the acrylics disk of thickness t. Acrylics was chosen as the mass made the sandwiches of the Langevin type structure because the acrylics did Langevin type resonance length short since acoustic velocity is lower than metal. Fig.9 shows the frequency characteristic of transmitting power sensitivity when changing the resonance length L (mm) of the Langevin type transducer. As a result of measuring resonance frequency about the four resonance length L= 26, 30, 36, and 40 mm, fr=46, 40, 30, and 20 kHz were obtained. Q factors at the time of these resonance were Q= 6.6, 6.7, 5.5 and 3.7, respectively, and all were low Q values. Although Fig.10 showed resonance frequency and the relation of resonance length, product Lxfr (this is called frequency constant) of resonance length and resonance frequency was 1500. From this result, in order to obtain broadband as much as possible, it was referred to as L=40 mm, and broadband speaker was made as an experiment using four layered piezoelectric devices. As a result, the frequency characteristic as shown by Fig.11 was acquired. Although compared also with the theoretical calculation result searched for the foregoing paragraph, it turns out that it is well in agreement. However, even if sensitivity fell rapidly with a low frequency band of 20 kHz or less and the frequency characteristic of trial production speaker enlarged the resonance length L of the Langevin transducer more, the sensitivity of a low-frequency region has not improved. Then, in order to improve a low frequency region, speaker only for a low frequency band was prepared, and it considered compounding with this proto type transducer. It is the actuator that has resonance frequency in the low frequency wave side using the mist beam layered piezoelectric device as speaker only for a low frequency wave band to 150 kHz is settled in less than 10 dB, ultra wideband transducer from 7 kHz to 150 kHz is realizable. Fig.12 shows the frequency characteristic that compounded transducer of low and a high region. Ultra wideband has been realized by a sensitivity deviation of less than 12 dB in the frequency range (7 kHz - 150 kHz).

4. Evaluation of ultra wideband speaker
In order to evaluate the ultra wideband speaker made as an experiment, the chirp waveform as shown in Fig.13 was transmitted from the ultra wideband speaker, it was received by the broadband hydrophone, and the frequency characteristic of this speaker was measured. Since it could not measure in the place that has multiple reflections, such as a tank, since the pulse width that transmits is long, it came out and went to the ocean. The dolphin speaker was installed downward so that it might not be subject to the influence of multiple reflection, and the hydrophone has been arranged directly under the speaker. An input signal is a linear FM signal. Transmitting sound pressure is kept from exceeding 180 dBuPa at 1m by the maximum sound pressure. The dolphin speaker of a prototype is shown in Fig.14. The signal transmitted from the speaker is recorded on data storage at the same time it is received by a broadband hydrophone and observed with an oscilloscope through broadband amplifier. It transmits from the speaker that made the linear FM signal (5 kHz - 170 kHz) as an experiment, and the result of having received it by the broadband hydrophone is shown in Fig.15. The transmitter pulse form from a broadband speaker was a deviation of less than 10 dB in the frequency range (8 kHz - 150 kHz). This attained less than ±6 dB made into the target.

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BASIC STUDY OF RHOMBOIDAL ACOUSTIC LENS CONSTRUCTED WITH PHONONIC CRYSTAL

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Abstract: A high-resolution acoustic imaging system is an important aid in turbid water where an optical camera system fails. Recently, phononic crystal (PhC) is synthetic materials that are formed by periodic variation of the acoustic properties of the material. Planar ultrasonic lens constructed with PhC that have a negative refractive index for ultrasound waves was designed by Dr. Zhang. In previous study, we fabricated a prototype of a planar acoustic lens made from stainless steel rods. However, the focal gain of the fabricated planar acoustic with PhC was lower than conventional lens, because the planar acoustic lens with negative refractive index allowed focusing of a sound when a spherical wave was incident to the plate. To fabricate a high gain lens constructed with PhC, we design a rhomboidal lens by focusing plane wave. Therefore, it is necessary to measure the dependence of negative refractive index at the plate constructed with PhC by incident angle of plane wave. In previous study, we measured the dependence of negative refractive index.

In this paper, we design a rhomboidal acoustic lens. In order to obtain basic properties of the lens, we simulated sound field focusing by rhomboidal acoustic lens with PhC by numerical analysis method. As a result, it is clearly shown that rhomboidal acoustic lens has a very narrow beam at f = 700 kHz. The rhomboidal PhC lens has a higher gain about 7 dB than the planar PhC acoustic lens at f = 700 kHz and it has a very narrow beam.

Keywords: phononic crystal, rhomboidal acoustic lens, numerical method, frequency characteristics of lens gain
1. INTRODUCTION

A high-resolution acoustic imaging system is an important aid in turbid water where an optical camera system fails. Many scientists have studied acoustic lenses in the past. The imaging system with acoustic lens was applied in the research of oceanic field [1]. Recently, phononic crystal (PhC) is very hot topics in ultrasonic fields. PhC is synthetic materials that are formed by periodic variation of the acoustic properties of the material. A phononic crystal is actually the sonic version of a photonic crystal. When propagating through periodic structures, sound waves reveal a particularly important property: propagation of the sound waves is modulated by the periodic structures. Planar ultrasonic lens constructed with PhC that have a negative refractive index for ultrasound waves was designed by Dr. Zhang [2]. In previous study, authors have been reported the basic properties of the planar acoustic lens for the development of imaging system with acoustic technology in ocean [3]. We fabricated a prototype of a planar acoustic lens made from stainless steel rods. For comparison between measurement and calculation results, we measured sound field focusing using the planar acoustic lens. Measurement results were in good agreement with simulation results. However, the focal gain of the fabricated planar acoustic with PhC was lower than conventional lens, because the planar acoustic lens with negative refractive index allowed focusing of a sound when a spherical wave was incident to the plate. To fabricate a high gain lens constructed with PhC, we design the rhomboidal lens by focusing plane wave. Therefore, it is necessary to measure the dependence of negative refractive index at the plate constructed with PhC by incident angle of plane wave. In this paper, to determine the dependence of refractive index at the plate constructed with PhC by incident angle of plane wave, we measured the direction of sound propagation getting through the plate which was rotated from 0 degree to 30 degrees. And, we design a rhomboidal acoustic lens. We reported the basic property of the rhomboidal acoustic lens for the development of imaging system with acoustic technology in ocean.

2. SIMULATION OF FOCUSING FIELD BY RHOMBOIDAL ACOUSTIC LENS CONSTRUCTED WITH PHC

Figure 1 shows the configuration of the PhC rhomboidal acoustic lens. The lens consists of triangular lattices. The PhC is made of stainless steel rods with a diameter and in-between interval of $a = 0.7 \text{ mm}$ and $d = 1.5 \text{ mm}$, respectively. The number of PhC layers propagating in the $z$ direction is thirteen. Pure water is around the rods. The cut angle of the front and back plane of the rhomboidal acoustic lens is $\theta = 30^\circ$. In previous study, we measured the dependence of negative refractive index. The negative refractive index of rhomboidal acoustic lens in case of $\theta= 30^\circ$ is $-0.56$ [4].

![Fig. 1: Configuration of the phononic crystal (PhC) rhomboidal acoustic lens.](image-url)
simulated the focusing properties of the rhomboidal acoustic lens by using the elastic finite difference time-domain method.

3. SIMULATION RESULTS OF SOUND FIELD FOCUSING BY RHOMBOIDAL ACOUSTIC LENS

To improve the performance of the rhomboidal acoustic lens, we simulated the focusing properties of the lens as a function of diameter of the rod in the PhC. Figures 2 (a)–(d) show the simulation results for the sound field in the z–x plane at frequency $f = 700$ kHz when $a$ changes from 0.5 mm to 1.3 mm. As shown in Fig. 2(a), for $a = 0.5$ mm, the propagated sound through the lens was not focused by the rhomboidal acoustic lens. Sound was spread in the direction of slant. As shown in Figs. 2 (b)–(d), the focal spot appeared in the sound field in the back of the lens. However, as shown in Figs. 2 (c) and 2 (d), a high-level side lobe exits near the focal point. To determine the focusing properties of the lens in each case, we show the sound pressure distribution on axis in Fig. 3. The sound pressure in this figure was normalized at the maximum value of each result. Figure

![Images of sound field focusing results]

*Fig. 2: Focusing the sound pressure field by changing the rod diameter a.*
3 (a) shows the sound pressure distribution in the propagation direction $z$. The focal length for equal to 0.7 mm and 0.9 mm was approximately 7 mm. The focal length shortens when $a$ increases. Figure 3 (b) shows the sound pressure distribution in the transverse direction $x$. The width of a $-3$ dB beam at the focal point was approximately 2.0 mm when $a$ changed from 0.5 mm to 1.3 mm. However, high-level side robe are generated for $a = 0.9$ mm, 1.1 mm, and 1.3 mm. On the other hand, for $a = 0.7$ mm, the sound level of the side lobe is approximately 7 dB lower than the rest of the results. Considering the amplitude ratio of the main and a side lobe, the PhC rhomboidal acoustic lens exhibits the highest performance for $a = 0.7$ mm.

4. FUNDAMENTAL FREQUENCY CHARACTERISTICS OF RHOMBOIDAL ACOUSTIC LENS

The PhC acoustic lens is a narrow band lens. To determine the frequency range, which can be used for focusing sound, the fundamental frequency characteristics were investigated by numerical analysis. To obtain the basic characteristics of the rhomboidal acoustic lens, we simulated the sound pressure distribution in the $z$-$x$ plane when the frequency changes from 500 kHz to 900 kHz. Figures 4 (a)–(d) show the sound pressure simulation results at frequency $f = 500$, 600, 700, and 800 kHz, and focus on the back of the lens sound field at $f = 700$ kHz and 800 kHz. On the other hand, a focal spot did not appear at the sound field back of the lens at $f = 500$ kHz and 600 kHz. The sound wave attenuated, and the sound pressure at the side of the axis is larger than the central axis at $f = 500$ kHz. Figure 5 shows the axial sound pressure distribution when the $f$ changes from 500 kHz to 900 kHz. The sound pressure in this figure was normalized at the maximum value of each result. Figure 5 (a) shows the sound pressure distribution at the propagation direction $z$. The focal length from the edge at the second plane of the lens is 7.2 mm and 34.0 mm at 700 kHz and 900 kHz, respectively. The focal length decreases when the frequency increases; however, the focal depth increases. Figure 5 (b) shows the sound pressure distribution of the focal point at the transverse direction $x$. The $-3$ dB beam width at the transverse direction $x$ is approximately 2.0 mm at $f = 700$ kHz and 800 kHz. The rhomboidal acoustic lens has a very narrow beam at $f = 700$ kHz. Especially, the lens has the characteristic that the beam width narrows when the frequency is low.
Fig. 4: Frequency characteristics of the sound pressure field focusing by the phononic crystal (PhC) rhomboidal acoustic lens.

Fig. 5: Sound pressure distribution on the z and x axis when the frequency f ranges from 500 kHz to 900 kHz.
To compare the lens performance between the rhomboidal acoustic lens and planar acoustic lens, we looked at the frequency characteristics of the lens gain. These results are shown in Fig. 6. Figure 6 shows the results for the rhomboidal acoustic lens and the results of the planar acoustic lens, which has the same acoustic parameters and triangular lattices. At 0 dB, these results define the amplitude of the sound wave at the focal point without a lens. As shown in Fig. 6, the maximum gain in the case of the rhomboidal acoustic lens is approximately 20 dB at \( f = 700 \) kHz. The gain of the lens gradually changes as \( f \) changes. The maximum gain in the case of planar acoustic lens is approximately 13 dB at \( f = 700 \) kHz. As a result, the rhomboidal PhC lens has a gain of approximately 7 dB higher than the planar PhC acoustic lens.

5. CONCLUSIONS

We discussed the design of a PhC rhomboidal lens. The rhomboidal acoustic lens has a very narrow beam at \( f = 700 \) kHz. Furthermore, the rhomboidal PhC lens has a gain of approximately 7 dB higher than the planar PhC acoustic lens.

In the future, the performance of the rhomboidal lens for designing an asymmetrical rhomboidal lens will be optimized by using genetic algorithms. We will fabricate a prototype rhomboidal lens using stainless steel rods. To confirm the performance of the rhomboidal lens, we plan to compare measurements and simulation results.

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REFERENCES

Session 9

Calibration of Sonar and Hydrophones

Organizers: Bo Lövgren and Stefan Schael
LOW FREQUENCY TANK CALIBRATION BY COMPARISON

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Abstract: In order to check the performance of Low Frequency Sonar Systems (1 kHz and below) there is a need for Calibrated Sources that can be arranged at different distances and in different environments. For that purpose Saab has developed a Signature Generation Module that fits in an AUV, and makes these performance checks easy to perform with low personal effort. In order to calibrate the source itself, it would be feasible to use a medium-sized test tank, but because of the small dimensions in relation to the wave-length there will be errors due to multi-path. This paper describes a method where the Source Level of one Calibrated Source is first determined in absolute terms in free-water, whereafter it is checked in the indoor Test Tank using a calibrated hydrophone. The difference in Estimated Source Level is then used for characterization of the Test Tank. This means that subsequent Calibrated Sources of the same design will only need to be checked in the Test Tank, in order to estimate the True Source Level. This paper shows an example of such a calibration scheme, together with the corresponding results and a discussion around possible sources of error.

Keywords: Test tank, low frequency calibration.
1. INTRODUCTION

In order to make accurate Source Level measurements below 1 kHz in a small undamped Test Tank, the conventional method of gating is not useful by (at least) two reasons.

- The difference in travel time between the direct wave and the first reflection is smaller than the wave period of the signal.
- The settling time for the transducer is larger than the wave period of the signal.

For a modular test tank as that shown in Fig. 1 (6 m x 4 m, depth 3.5 m), with transducer and hydrophone mid-depth – with transducer in the centre and hydrophone 1 m from the short side – the travel time difference is about 1.33 ms, corresponding to 750 Hz.

![Fig. 1: Example of a Low Frequency Transducer, lowered in the Test Tank.](image)

2. THE EFFECTS OF MULTI-PATH

The effects of multi-path when trying to estimate the Source Level from a Low Frequency Transducer are shown in Fig. 2, where a Free Flooded Ring Transducer is excited with a Continuous Random Noise signal in the range 0 to 1.6 kHz.

Comparison to the Transmit Graph delivered by the manufacturer shows a number of additional features which are due to the multi-path phenomena, which lead to a complex pattern of constructive and destructive interference.

Even if the transducer used in the example isn’t very efficient below 1 kHz, the effects of multipath are visible anyway. Because of the need for a continuous signal, in order for achieving sufficient averaging during the spectrum analysis, multi-path effects would appear even during free-field conditions – by practical reasons. This is shown from a recalculation of the Lloyd-mirror anomaly resulting from the simplified Test Setup shown in Fig. 3, arranged with the Test Ship UW3 steadily anchored.
Fig. 2: The effects of multi-path in a Test Tank when using Random Noise Excitation.

Fig. 3: Test Setup for free-field Source Level measurement.

In this example, the following test parameters are assumed:
• Water depth $D_W = 40$ m (flat bottom, 50 % reflectivity).
• Hydrophone and Test Object depth $D_H = D_{TO} = 20$ m.
• Hydrophone to Test Object distance $R = 15$ m.
• Surface/Ship reflectivity 80 % (with 180° phase shift).

The resulting Lloyd-mirror anomaly (i.e. Transmission Loss difference in dB from spherical spreading $20 \log R$) is shown in Fig. 4 together with the effect of changing the hydrophone depth $D_H$ to 19 m and 21 m as well.

In this simple example, the regular pattern of constructive and destructive interference is obvious. Because of the fact that the received levels of the Reflections are less than that of the Direct Path, the pattern is almost symmetric around the 0 dB level. It could also be assumed that each of these zeroes appear almost exactly between a maximum and the subsequent minimum, or vice-versa. If the spectrum of the transducer Source Level is smooth, these “zero frequencies” can then be used as sampling points for the True Source Level estimation.

The geometric symmetry also leads to constant values for the max and min values for the anomaly when $D_H = 20$ m. When changing $D_H$ to 19 m and 21 m, the pattern changes due to the changing phase shifts caused by the changes in travel-time. One effect of this is that the frequencies for 0 dB anomaly changes. Measuring at several different depths thus causes the total “density” of these zeroes to increase, at least in certain frequency ranges (such as between 300 Hz and 600 Hz in Fig. 4), leading to an increased number of possible sample points for the True Source Level.

Even if the anomaly emergent in the Test Tank environment is much more complex than that shown in Fig. 4, it could still be assumed that there will show up maxima and minima in that situation as well. And if the level variations of the anomaly are small, the
“zero level anomaly” could be applied even here, though at different frequency values than during the free-field measurements.

3. A METHOD FOR SOURCE LEVEL CALIBRATION

Fig. 5 shows the resulting Frequency Spectrum for the received signal from the Free Flooded Ring Transducer after moving the hydrophone horizontally, compared to that shown in Fig. 2. In the same way as in Fig. 4 the local maxima and minima appearing around the assumedly smooth Frequency Characteristic of the transducer has moved in frequency and amplitude related to those in Fig. 2.

![Fig. 5: Test Tank measurement with changed geometry.](image)

Even if the interference pattern will be more complex in the Test Tank, it could anyway be assumed that it is relatively stable from time to time for the same type of transducer (or other Test Object) of the same kind, given that the Test Geometry is kept the same. And by moving the hydrophone slightly, as in Fig. 5, it can be shown that the average levels over sufficiently wide frequency bands B (in this case 200 Hz, indicated by asterisks) keep quite stable. This makes these averages useful for comparison.

This enables the following Test Regime to be applied for the Acoustic Calibration of a series of otherwise equal Test Objects, outgoing from one single Free Field measurement campaign.

**True Source Level estimation for Test Object #1**

1. Measure the Hydrophone Output Level in Free Field-conditions with Band Limited Random Noise as input to Test Object #1 for one or more Test Geometries.
2. Estimate the frequencies for the local max and min levels in the Frequency Spectrum of the Hydrophone Signal.
3. Estimate the frequencies for null anomaly from the max/min data.
4. Use Sinusoidal input at the null anomaly frequencies to establish values for the True Source Level $SL_{T1}$ by means of the distance $R_{T1}$, the Hydrophone Sensitivity $S_{H0}$ and the corresponding Hydrophone Output Levels $L_{HT1}$ (1).

$$SL_{T1} = LH_{T1} - S_{H0} + 20\log R_{T1} \text{ [dB re 1 µPa @ 1 m]}$$  \hspace{1cm} (1)

**Equivalent Source Level estimation for Test Object #1**

1. Measure the Hydrophone Output Level in Test Tank-conditions with Band Limited Random Noise as input to Test Object #1 for one suitable Test Geometry.
2. Estimate the average over B Hz wide bands to establish values for the Equivalent Source Level $SL_{E1B}$ by means of the distance $R_{E1}$, the Hydrophone Sensitivity $S_{H1}$ and the corresponding Hydrophone Output Levels $L_{HE1B}$ (2).
3. Interpolate the Free Field results in order to estimate the corresponding True Source Levels averages over B Hz wide bands $SL_{T1B}$.
4. Compare the Free Field results and the Test Tank results to establish the Calibration Function $F_{TEB}$ given by (3).

$$SL_{E1} = LH_{E1} - S_{H1} + 20\log R_{E1} \text{ [dB re 1 µPa @ 1 m in B Hz bands]}$$ \hspace{1cm} (2)

$$F_{TEB} = SL_{T1B} - SL_{E1B} \text{ [dB]}$$ \hspace{1cm} (3)

**True Source Level estimation for Test Object #n**

1. Measure the Hydrophone Output Level in Test Tank-conditions with Band Limited Random Noise as input to Test Object #n for the same Test Geometry as for #1.
2. Estimate the average over B Hz wide bands to establish values for the Equivalent Source Level $SL_{EnB}$ by means of the distance $R_{E2}$, the Hydrophone Sensitivity $S_{Hn}$ and the corresponding Hydrophone Output Levels $L_{HEnB}$ (4).
3. Interpolate the Free Field results in order to estimate the corresponding True Source Levels averages over B Hz wide bands $SL_{TnB}$.
4. Insert $SL_{EnB}$ in (3) to estimate the True Source $SL_{TnB}$ by reverting (3) to (5).

$$SL_{EnB} = LH_{EnB} - S_{Hn} + 20\log R_{En} \text{ [dB re 1 µPa @ 1 m in B Hz bands]}$$ \hspace{1cm} (4)

$$SL_{TnB} = SL_{EnB} + F_{TEB} \text{ [dB re 1 µPa @ 1 m in B Hz bands]}$$ \hspace{1cm} (5)

4. **POSSIBLE SOURCES OF ERROR**

There are three distinctive areas of error that can affect the accuracy of the subsequent estimations of the True Source Level, $SL_{TnB}$. The first comes from the of the initial Free
Field measurement results, LH_{T1}, the second from the Test Tank measurement results, LH_{EnB}, and the third from the different recalculations using formulae (1) to (5). Below are shown some of the facts that have to be taken in consideration for a deeper error analysis.

**Free Field measurement errors**

- Rapid time fluctuations of the path lengths causing amplitude variations that affect time averaging of the Hydrophone Signal.
- Slow time fluctuations of the Surface Reflectivity, which can change the interference pattern, exemplified in Fig. 4.
- The rapid change of level versus frequency at the frequencies of zero anomaly, resulting in possible errors in level.

**Test Tank measurement errors**

- The complex interference pattern makes it difficult to find a proper measurement geometry that leads to a low sensitivity for positioning errors at the comparison.
- The larger variations in level due to multipath, in comparison to the Free Field case, leads to a higher risk for averaging errors.

**Recalculation errors**

- The number of steps included will undoubtably lead to a greater inaccuracy for the recalculated True Source Levels, SL_{TnB}, than for the really measured one, SL_{T1}.
- The major additional errors come from the range estimates, R, which become more important in the Test Tank measurement case, where the relative errors will be larger than for the Free Field measurements.
- In the Test Tank measurement case it is therefore important to keep the measurement geometry as equal as possible between comparisons by means than could possibly make R_{En} almost equal to R_{E1}.
- The same goes for the Measurement Hydrophone, where it would be worthwhile to use exactly the same one for each comparison, which also makes S_{Hn} and S_{H1} equal.

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ADULTERATION OF UNDERWATER ACOUSTIC MEASUREMENTS

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Abstract: The awareness of underwater threat will always be a constant companion of the NAVY. All efforts have been done to analyze and minimize the unknown threat and will be continued in the future.

The underwater emission generated by different components and hydrodynamic behavior are evaluated by a hydro acoustic measurement system. Procedures and recommendations for silencing the underwater signature are dependent on the accuracy of the range facilities. The influences caused by the environmental area, uncertainties of the steady state of the ship and the technical range system have to be taken into account.

The German subordinated Technical Center for Ships, Naval Weapons Maritime Technology and Research (WTD71) of the Federal Office of Bundeswehr Equipment, Information Technology and In-Service Support operates a fixed underwater signature range in shallow water and in cooperation with Norway and the Netherlands a fixed underwater acoustic range in deep water. In non-routine cases, a mobile measurement system will be deployed in shallow water areas. The development and installation, including the quality acceptance tests of those measurement systems, are carried out by WTD 71, which is accredited to perform the calibration of hydrophones. Periodical inspections and calibrations assure that the high quality demands are met.

The quality of the electronic measurement chain depends on the calibration of the sensors and the installed components. A short overview of the measurement chain will present the components in use. The procedure of the system calibration will be shown as well as the comparison of different international mobile range systems, measuring the same sound source at the same location.

Keywords: hydrophone calibration, mobile underwater measurement system, comparison
1. Introduction

The possibilities of global trade, international merchandising and shifting the productions lines to foreign countries subsequently lead to the increase of international shipping traffic lanes. The influence of the marine environment caused by radiated sound of the ships has become in the focus of research and surveillance programs. A multitude of groups are deploying sensors systems in the corresponding areas. The demand on these systems and sensors reflects the prospective market situation for the industry. In addition to the established companies, new suppliers are entering the market with new hydrophones or underwater acoustic systems. The individual intention of the user adapts those COTS (commercial of the self)-systems to their special needs. Regarding the diverse influences on each measurement campaign the results of suitable and repeatable performances could vary enormously according to the used measurement- and analysis procedure, the local and time dependent environment, the distance sensor to ship, aspect angle and of course to the properties of the used technical system. Based on these impacts the outcome of corresponding research and surveillance programs might lead to discussions in the international forum.

International environmental groups have initiated a Marine Environment Protection Commitee within the IMO (International Maritime Organization) to reduce shipping noise. Identified as a prerequisite, a standard for measurement has to be released. In a joint working group, within the ISO (International Organization for Standardization) efforts and demands will be harmonized to establish standardization for ship measurements.

Gaining results by one underwater acoustic measurement system or another one could experience a huge difference as the following project will reveal.

2. Objective

In the international project SIRAMIS (Signature Response Analysis on Multi Influence Sensors) coordinated by the European Defence Agency seven nations agreed to investigate the merchant ship underwater signatures related to the ship own parameter and the interaction with the sensors in a realistic environment. The WTD 71 was requested to conduct a benchmark trial to compare the different international mobile underwater measurement systems in reference to their underwater signature range.
3. Calibration of the underwater measurement system

The underwater acoustic measurement of the radiated noise of ships will be impacted by manifold causes e.g. the configuration of the sailing ship, the local environment and the measurement chain itself.

Considering these influences, one should start to mitigate the impact on the measurement by a system with an unknown or uncertain transmission response. Several standardized test procedures have to be performed to obtain the correct sensitivity values regarding the measurement chain.

Comparing results of relevant measurements are only qualified by a calibrated measurement chain.

The WTD 71 has been operating a fixed installed underwater acoustic measurement range since the early 60’s. The first and main reason is to measure the radiated noise of new ships and evaluate the results as stipulated in the contract. Today, with its continuous ongoing improvement and experience, WTD 71 operates one of the world wide most unique underwater acoustic range.

3.1. Measurement Chain

In general, an underwater acoustic measurement chain is comprised of several components. Transferring the registered signals of the hydrophone to the analyzer or recorder includes at least a signal conditioning and depending on the type of deployment a long cable connection as transfer path. The signal conditioning will adjust the electronic of the sensor and connecting devices, preamplifies the signals and or convert them. On base of the chain illustrated in the figure 1 the system calibration will be descripted.

![Outline of an underwater acoustic measurement chain](image)

*Figure 1 Outline of an underwater acoustic measurement chain. Blue box captured items are underwater*

At the wet end of an underwater measurement system we find the most sensitive component, the hydrophone with its own properties of the conversion of mechanical force (pressure) into
electrical values. The mostly used sensor material for hydrophones is based on the polycrystalline ferroelectric ceramic, e.g. PZT (lead zirconate titanate). This anisotropic raw material shows higher induced amplitudes than mono-crystalline material, e.g. quartz. Special doping of this raw material will optimize the field of application. A sufficient piezoelectric effect will be achieved by the polarity process. An equivalent electrical circuit of the sensor delivers the theoretical frequency response and the sensitivity of the sensor. Hydrophones are designed with diverse built in components which will differ from this assumption. Starting in the low frequency range the circuit describes a high pass filter with \( f_g \approx 1-7 \) Hz (depending on the capacitor in the hydrophone) while in the high frequency area the oscillating circuit appears as a low pass filter with \( f_g \approx 60-160 \) kHz. The components of the electric circuit are dual elements of the piezoelectric materials. The electrical connection to the sensor is taken into account by the input resistance \( R \) and capacity \( C \) in the electric circuit (shown in Figure 2).

![Figure 2 Sensitivity level over frequency of the equivalent circuit of the hydrophone](image)

Due to the effects of depolarization throughout the life cycle of the material and the inaccuracy of the electrical components the hydrophones will be calibrated at the beginning by the manufacturer and during the life cycle in responsibility of the user.

### 3.2. Sensor calibration

The operational functionality of an underwater acoustic range with its regarding quality assurance includes the periodically calibration of the hydrophones at an accredited facility as well as the system evaluation of the measurement chain deployed together with the calibrated hydrophone.

The wide frequency range requires different calibration methods. Low frequency range, up to 2 kHz, will be tested in air since the frequency response is nearly constant in this range. The calibration of hydrophones in the higher frequency range can be performed in different procedures. The simplest procedure is to compare the results with a calibrated hydrophone regarding the same geometry, environment, directivity and transducer (Substitution method).
At least one reversible transducer will be used in the reciprocity calibration. This method is implemented in the procedure of the WTD 71.

![Figure 3 scheme of layout for the hydrophone reciprocity calibration](image1)

\[ M_{R1}(\text{receiving sensitivity}) = \sqrt{\frac{U_{R1}}{U_{R1/S}} \cdot \frac{2d}{U_{R} i_s \cdot f \rho}} \]

*Figure 3 scheme of layout for the hydrophone reciprocity calibration*

The calibration of the sensor itself follows the rules of the DIN EN 60565 *Underwater acoustics- Hydrophones-Calibration in the frequency range 0.01 Hz to 1MHz*.

Instead of using a water tank, the WTD 71 established an accredited facility in a lake, the Plöner lake. According to the wave length, the free field conditions in this lake are limited by the geometry of the mechanical arrangement holding the transducer and hydrophones and by the foundation of the laboratory which is positioned in the lake on piles.

![Figure 4 Diagram of the calibration results of one hydrophone over the life cycle](image2)

*Figure 4 Diagram of the calibration results of one hydrophone over the life cycle*

As an example, figure 4 illustrates the results of the reciprocity calibration of one hydrophone over a time period of 10 years. It shows variations less than 1.5 dB which is within the uncertainty range of the underwater measurement. Therefore, this hydrophone did not change the frequency dependent sensitivity. In case of a wide spread or a tonal deviation the hydrophone would not be used further in the measurement chain. New hydrophones will be calibrated at this facility as well and compared with the manufacturer calibration.
3.3. Frequency response of the transfer path

In general, the underwater acoustic measurement system should be designed for a sufficient signal to noise ratio. Therefore, the cable termination should provide a high resistance. The next components in our measurement chain (Figure 1) are a signal conditioning unit, cables and the amplifier which have to be reviewed. Electrical properties occur over this end and have to be evaluated in a measurement. Considering these properties, the electrical response will be retrieved at the 1/3 octave band mid frequencies. Most hydrophone types are equipped with an internal amplifier. Incorporating the influences of this amplifier, additional measurements with the hydrophone in the loop are performed in the frequency range up to 100 Hz. Referring these results to 100 Hz, the values of the measurement without the hydrophone in the loop will be corrected. It is necessary to include the amplifier at the end of the chain (Figure 1) according to its non-linearity in the high frequency bands. The system calibration is applied for the analog transfer path and is carried out by voltage reference measurements. After the analog/digital conversion, only quality tested processes using calibrated generators and multi-meter are performed. Figure 5 reveals the influence of the built in amplifier in the hydrophone as well as the non-linearity of the amplifier, otherwise the frequency response is constant.

![Figure 5 frequency response measurement chain](image)

Adding the frequency response of the measurement chain to the calibrated sensitivity of the hydrophone, the raw measurement data will be evaluated by the total sensitivity values in the processing unit.

4. Comparison of different UW measurement systems at one location with the same sources

The deviation, by using different measurement systems for the evaluation of the radiated noise of a ship, should be negligible. Especially in case of contracted radiated noise limitations for ships, approved by the international community, uncertainties in the technical systems should be minimized. In the scope of the SIRAMIS project different international mobile underwater measurement systems had to be compared. The deployment of these
systems took place at the underwater signature range of the WTD 71 in Aschau. The water depth is about 22 m. The mobile UW-systems were layout in one line, (track-line) in a distance up to 400m to the reference system. The layout can be seen in figure 6.

![Figure 6 Layout of the deployment of the Uw-mobile systems](image)

The ships sailed several times in the same operational mode over all systems. The aspect angle and the distance of ship/sensor varied with the track deviations. The controlled runs were tracked by DGPS. Runs with deviation over 5m off track were dismissed. The influence of the aspect angle could be neglected. The radiated noise level in this comparison was not corrected to 1m. Corrections by distance or amplifications will have an effect like an +/- offset. The calibration and frequency response measurement of the different systems were performed at different institutes.

![Figure 7 a) Comparison with 3 different sources for system I; b)-with two different sources system III](image)

The comparison is based on so called quick look results. The analyses are processed by formula 1 and corrected by the individual sensitivity. The time window \( t \) includes the ship length plus 2-3 sec after stern. Several runs of one configuration, analyzed by formula (1), were averaged for each system and applied as basis for the comparison.

\[
RNL(f) = \max_c OTO(f,t) \quad (1)
\]

\( OTO(\text{one Third Octave spectrum}); RNL(\text{Radiated Noise Level}) \)

The obvious deviation in Figure 7 (left) of one system in reference to the Aschau range have to be distinguished in low and higher frequencies. For three different sources (ships) the
differences between the reference hydrophone and the mobile system tend to the same deviation. The huge significant deviation in the low frequency range up to 31 Hz seems to be caused by the sea bottom. The Eckernförder Bay, in which the underwater range is located, has a muddy and gassy sea bottom. The influence of the bottom properties at the reference hydrophone is prominent. Further investigations will explain the different effect in 400m distance to the references system, which was measured with the mobiles systems. The variation in the frequency range from 50Hz-20 kHz is most likely not originated by the sea bottom or by the source. The repeatability of the runs was within of 2-3 dB in this frequency range. The same shape of the deviation for three different sources looks like system immanent. The variation in this comparison was expected much lower. Results, gained by the two other mobile systems, vary in a different way which excludes possible causes by the reference system. Detailed analyses and investigations are ongoing within the international project to explain the differences.

5. Conclusion

There are several effects which will influence an underwater measurement result. Besides the environment, aspect angle, distance sensor –source and the source configurations, the measurement system itself will influence the results. The calibration charts reveals the stable sensitivity over the life cycle. Deviations in an order of more than 2% are not acceptable. Connecting the hydrophone to the measurement chain could change the sensitivity in the low frequency bands and dependent of an amplifier also in the high frequency bands. Experienced knowledge and the possibility to compare several hydrophones at one location by using a controlled source, procedure within WTD 71, exclude hydrophones with an unusual behavior. The comparison of different underwater measurement systems within the project SIRAMIS at the same location and by using the same controlled sources shows higher deviation than expected. Different calibration method at different institutes as well as system immanent properties could be an explanation. In the low frequency band the sea bottom has the most prominent influence. Further investigation will be performed.

References


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THE CALIBRATION OF HYDROACOUSTIC CHANNEL
OF MOBILE MEASUREMENT MODULE

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Abstract: In recent years, remarkable interest was noted down in range research area connected with topic: observance the same or different systems with the same declared parameters for measuring ship signatures at different sea regions for different environmental and hydro-meteorological conditions. It's particularly related to measuring the acoustic signature of the ship. The article describes a detailed procedure for calibrating hydroacoustic measurement channel of mobile module made by the Polish Naval Academy. The calibration chain consists of: hydroacoustic transducer with elements of electronics and acoustic channel with analog-to-digital converter and signal processing procedures using time domain and one-third octave (OTO) analysis. Calibration takes into account test insert voltage by specifying a sinusoidal signal with constant amplitude with center frequencies in the band OTO from 3 Hz to 25 kHz. The results of the calibration tests are characterized in the article. Performing this calibration procedure it was necessary to compare the results obtained with the use of acoustic measurement systems with other participants in the study. These studies, which will provide knowledge on ship signatures collected using the systems of the various participants in different sea areas under diversified environmental conditions, were carried out within the SIRAMIS project under the European Defense Agency.

Keywords: hydrophone calibration, system comparison
1. INTRODUCTION

Hydroacoustic field measuring circuit is a multi-component system [1]. It consists of a sensor, as a hydrophone and a recorder with analogue to digital converter.

Each measuring system requires regular inspection and calibration operation. This operation eliminates any constant errors affecting the correctness of the measurements results. Reliable calibration demands that each element, as a part of a complete measurement circuit, has to be checked. It is calibrated independently from each sensor element to the final stage, which extracts data processing results. What is more, definitively independent control should to be done, to compare its results with the results of the entire channel.

This article, firstly will explain how to verify the proper operation and calibration of the measurement path, and secondly will present test results and comparison with previous calibration results.

2. COMPARISON OF HYDROPHONE CALIBRATION RESULTS

Measurements were conducted with hydrophone Reson TC-4032, operating in usable bandwidth range to 120 kHz. The specific hydrophone was provided all together with its directivity pattern and relations of voltage sensitivity to frequencies response. We took into consideration average value of voltage sensitivity of measuring bandwidth to signal processing. This value was based on attached diagrams. Basis of the value was due to attached diagrams. The average value in the band up to 25 kHz was about -171 dB re 1V/μPa.

Regarding the hydroacoustic field results of measurements differences between all measurement modules there was specified direction of further actions. The first step was to recalibrate by the manufacturer the hydrophone used in measurements. Calibration was conducted in Reson measurement laboratory. The calibration results showed that the average sensitivity in the analyzed band increased by 1.4 dB re 1μPa. Detailed differences are shown in Figure 2 [2, 3].

Fig. 1: Comparison of voltage sensitivities of hydrophone Reson TC-4032.
3. INSERT VOLTAGE CALIBRATION

The next step, which allowed comparing different measurement systems, was to verify appropriation of the signal processing circuit. The device with integrated FPGA and real-time system with the analog-digital converter was used to record the specified parameters. Parameters of converter were selected in accordance with the requirements for system calibration. 24-bit AD converter allow for the signal recording of a wide dynamic range from microvolt’s to few volts. For input voltage range from -5V to 5V voltage resolution was approximately 600nV. Sampling frequency 51.2 kHz enabled recording useful signals from DC to 25 kHz. The recorder worked supported by software dedicated for device management, measurement card, fast FPGA system and Ethernet interface controlling. Another application installed on the host computer was responsible for the communication with the recorder, sending control messages, receiving and recording data on the hard disk. The software was implemented in LabVIEW environment due to the manufacturer of a complete recorder, National Instruments.

The correctness operation test of the measuring circuit from the recorder to the final stage of data recording to hard disk was carried out by attested function generator Agilent. The generator was connected instead of hydrophone to hydroacoustic channel (Fig. 3).
The test consists of going out sinusoidal signals obtaining from a generator to module input. Duration of each signal was 30 s and frequency as central frequencies of each one-third-octave from 3.15 Hz to 25 kHz. A difference between the signal going out directly from the generator and signal recorded by the module is shown below (Fig. 4).

![Fig. 4: Signal in time domain from generator and recorded by measurement module.](image)

It was assumed constant amplitude of 250 mV according to the voltage sensitivity of the hydrophone can be converted to the corresponding pressure.

For receiving the voltage sensitivity equal to $-172\text{dB re } V/\mu Pa$

$$-172\text{dB re } V/\mu Pa \Rightarrow 2.511 \text{ mV } /\text{ Pa}$$

$$P = \frac{\sqrt{2} \cdot 250 mV}{2.511 \text{ mV } /\text{ Pa}} = 70.37 \text{ Pa}$$

$$P(\text{dB}) = 20 \cdot \log_{10} \frac{70.37 \text{ Pa}}{1 \mu Pa} = 157 \text{ dB re } 1 \mu Pa$$

Results of comparison for signal recorded by the measurement module acoustic channel with standard pressure value for the center frequency bands of one-third-octave shown in Figure 5.

![Fig. 5: Pressure in frequency domain of based signal (blue) and recorded by module (red).](image)
This section shows an application window used to processing of the data recorded from four hydroacoustics channels. In the upper left window it’s shown the narrowband spectrum. In the upper right corner there is the RMS signal. In the lower left window one-third-octave spectrum of signal. The lower right window shows the signal in time domain.

![Window of post processing application.](image)

It is possible to obtain the most important information contained in signal thanks to application which processed recorded raw data. Furthermore, the processing results can be saved to a file in both graphical and numerical format. The outcome files were used eventually to compare results extracted with all the other teams’ measurement modules. Narrowband spectrum was gained using the frequency resolution of 1Hz with intervals of length equal 2s of signal with a 50% overlap. Spectrum range was between 3 Hz and 25 kHz. One-third-octave spectrum was obtained from the same range as narrowband spectrum.

The algorithms used to calculate the narrowband spectrum and one-third-octave spectrum are complete implemented algorithms in Sound & Vibration toolbox of development environment LabVIEW.

Comparison of the signal amplitude and the values obtained from one-third-octave spectra are shown in the following figure (Fig. 7). Summarizes the differences between the level of the signal from generator given to the hydrophone calibration input and the amplitude of the signal obtained from the recorder (blue line). The difference between the level of the signal generator and the one-third-octave spectrum level is shown as a red line. In the software there was implemented a third order band-pass filter covering the bandwidth corresponding to the assumed band of signal analysis.
4. CONCLUSION

Recalibrating the track measuring system revealed necessity cyclic control of measuring chain. The article shows that each of the components has a significant impact on the accuracy of the final display. In the case of mutual comparison of different systems indications, it is important to monitor both the sensitivity drift of the sensor with an amplifier (analog part of the system) and the digital part as an analog to digital converter along with signal processing procedures. In addition, these studies have shown necessity measure the sensor placed away from the structure of the module to verify the possible impact of the electronics housing containing a certain volume of air. What is planned for the near future.

REFERENCES

CALIBRATING HYDROPHONES AT VERY LOW FREQUENCIES

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Abstract: Low frequency hydrophones are a cost effective way to augment seismometers. However facilities for calibrating hydrophones at very low frequencies are not readily available. This paper describes the design and initial results of the Ocean Networks Canada VLF hydrophone calibration system.

Keywords: hydrophone calibration, very low frequency
BACKGROUND

Ocean Networks Canada (ONC) measurements of near-field earthquake energy occur primarily below 60 Hz. ONC measurements of far-field earthquake energy are typically below 2 Hz. The frequencies produced by turbidity currents are not known at this time.

Seismometers are the primary tool for studying these events, providing a time series of acceleration in three spatial directions. On land seismometers are expensive, fragile, and difficult to emplace. On the seafloor the issues are compounded. Broadband (BB) ocean seismometers, Fig.1, operating from 3 mHz to 100 Hz, cost the observatory roughly $200K in parts and over 100 man-hours in assembly and testing per seismometer. It requires a minimum of 1 full day of ship and ROV time to deploy at $75K per day. Short period seismometers are less costly to assemble at roughly $30K, but with a bandwidth of 1 to 100 Hz they are limited to near-field events.

Fig.1: BB Seismometer going into Caisson. Fig.2: icListen LF hydrophone tripod

Low frequency hydrophones, such as the icListen LF, shown in Fig.2 on an easy to deploy tripod, and the icListen AF are even less costly at $13k and require only about ½ hour of ship and ROV time to deploy. A VLF calibration can expand the manufacturer’s calibrated range down to 20 mHz or lower, as shown in Fig.3. With 24 bit digitizers these hydrophones have sufficient resolution to provide useful calibrated data at these frequencies.

Fig.3: Extending the calibration range of an LF hydrophone down to the 10’s of mHz

A comparison of a co-located BB seismometer and LF hydrophone is shown in Fig.4. Although the hydrophone saturated it shows that hydrophones are suitable for event detection systems and contributing to hypocentre determination. For the same cost as a
BB seismometer an array of hydrophones can be deployed which could improve automated early warning systems and provide data to allow inversion of geophysical parameters from near-field events.

![Time series data from seismometer (top) and LF hydrophone (bottom). Sensors were 250km away from magnitude 6.6 earthquake 24Apr2014.](image1)

Hydrophones should also be suited to near-field detection of underwater landslide and turbidity currents however this is a nascent field of research and could prove false.

There is much to be learned if we can measure the acoustic power spectral density of the events. To do this the hydrophones must be calibrated well below 1 Hz. Calibrations at these frequencies are not presently provided by hydrophone manufacturers or independent calibration facilities.

To meet this requirement ONC has developed a Very Low Frequency calibration capability in house. A prototype system has been operated for over a year. Recently a new system has been designed, manufactured and submitted for patent. The first calibration was run on this system in April 2014 on a Saab owned Reson 4032-1 analogue hydrophone.

**DESIGN DESCRIPTION**

The VLF calibration vessel is shown in *Fig. 5* and a section view of the vessel is shown in *Fig. 6*. A small volume of water or oil (A) is contained within a very rigid pressure vessel. The hydrophone element to be calibrated (B) is sealed in the liquid along with a
drive piston, temperature sensor, hydrostatic pressure sensor, and reference sensor. Both
the piston and reference sensor are pressure balanced in that the backside of both devices
are exposed to the same hydrostatic pressure as the test chamber. A slide valve allows the
system to be pressurized and isolates the backside of the reference sensor and piston
during the calibration.

![Fig.5: VLF Calibration Vessel.](image)

The reference sensor is a differential pressure sensor with a flat response from 0 to >2
kHz. The reference sensor output voltage is calibrated to pressure in Pa.

The 4 cm diameter piston is driven by a stack actuator with a maximum travel of 10
µm.

The internal dimensions of the main chamber (A) govern the maximum operating
frequency of the VLF calibration system. In order to obtain 0.1 dB accuracy between the
pressure seen at any point in the calibration chamber, the longest internal dimension
should not exceed 1/20th of a wavelength of the insonifying sinusoid. With a longest
internal dimension of 12 cm the chamber limits the maximum useable frequency of the
system to approximately 600 Hz. The maximum frequency limit is calculated by Eqn. 1
where \( c \) is the sound speed of the liquid, \( \lambda_{\text{min}} \) is the minimum allowable wavelength, \( d \) is
the maximum internal dimension, and \( E \) is the error limit in dB.

\[
 f_{\text{max}} = \frac{c}{\lambda_{\text{min}}} = \frac{c \cos^{-1}\left(10^{(-E/20)}\right)}{\pi d}
\]  

(1)

The calibration vessel is certified to a working pressure of 3000 dbar however the
design of many hydrophones limit calibrations to lower pressures. This can be due to the
depth rating of the hydrophone or limitations on the sealing of the hydrophone in the
calibration vessel. Presently the vessel has seals for the Ocean Sonics icListen LF and HF
hydrophones, Reson 4032 hydrophones, and the GeoSpectrum M8 hydrophones (the
GeoSpectrum hydrophones are used on the JASCO Applied Sciences tetrahedral arrays).
The hydrophone insertion port is 50 mm in diameter which limits the size of the hydrophones which can be calibrated.

The vessel is submersible to allow the system to be operated at temperatures from 1 to 40°C. This also facilitates thermal stability. Thermal stability is crucial during calibrations of LF hydrophones, such as the icListen LF, since these hydrophones are sensitive to pyroelectric variations. The user interface for the calibration system monitors the bath temperature and internal temperature of the chamber. The temperature differential limit for LF hydrophones is set to 0.1°C to prevent saturation of the hydrophone.

The user interface was developed in MATLAB 2012b and is designed to automate the calibration through all the user specified frequency points.

**METHOD**

The calibration method is similar to section 5.4 of the ANSI S1.20-1988 standard but differs in several key ways. The differences include the use of a differential pressure sensor as the reference, a stack actuator driven piston, internal and external temperature and hydrostatic pressure sensors, and a submersible high pressure test chamber.

The calibration vessel is filled with water or light mineral oil with care that no bubbles are trapped in the main chamber with the hydrophone. Bubbles of any size will reduce the generated acoustic pressure in the vessel dramatically. Additionally the bubble(s) create anti-resonant and resonant modes in the pressure vessel. Before a calibration point is taken the valve that isolates the main chamber, the backside chamber of the reference sensor, and the backside chamber of the piston is closed. The piston is then driven to insonify the main chamber. The reference sensor and hydrophone data are then captured simultaneously for 10 cycles of the insonifying frequency. The digitized reference pressure sensor voltage is converted to an rms pressure in Pa. The hydrophone output is converted to rms counts for digital hydrophones or rms voltage for analogue hydrophones. The sensitivity is reported in dB re counts²/μPa² for digital hydrophones and in the standard dB re Volts²/μPa² for analogue hydrophones.

**INITIAL RESULTS**

The first calibration performed with the new VLF calibration system was performed on a Saab owned Reson 4032-1 hydrophone. This calibration was completed from 1 kHz down to 2 Hz. The calibration was run in light mineral oil at 23°C with a Sound Pressure Level of 207 Pa or 166 dB. The calibration error of the reference sensor was 2 Pa peak over a range of ±1000 Pa. The sound speed in the mineral oil is approximately 1325 m/s resulting in the estimated chamber error shown in Fig. 7 as the red dashed line. The maximum reference digitizer error is 0.78% of reading. The combined error versus frequency is shown as the solid line in Fig. 7.

The initial calibration was hampered by a nick in three O-rings on the valve caused by a sharp port opening. This caused a small loss of pressure in the calibration chamber which prevented taking data points at frequencies below 2 Hz in the new calibration vessel.

The initial calibration also highlighted a problem with the compliance of the various O-rings in the new chamber. This problem was not seen in the prototype since all joints were epoxied tight. To reduce the O-ring compliance effects in the chamber it was necessary to
pressurize the chamber to at least 100 dbar to force the O-rings hard against the O-ring grooves. *Fig. 8* shows the sensitivity plot of the Reson 4032-1 hydrophone.

![Fig. 7: Error analysis (oil at 166 dB SPL).](image1)

![Fig. 8: Reson 4032-1 sensitivity.](image2)

**FUTURE WORK**

The VLF calibration system is presently designed to allow both digital and analogue hydrophones to be characterized in five dimensions. These dimensions include hydrophone output versus frequency, temperature, pressure, and sound pressure level. Output versus frequency is the only work that has been done to date with the new calibration system. Sound pressure level characterizations were performed using the prototype system but have yet to be done using the new system. Temperature and pressure characterizations have yet to be performed.

Mapping the phase versus frequency of the hydrophone has yet to be incorporated into the design. This is important for accurate time measurements when correlating signals between hydrophone models at frequencies below 1 Hz.

**ACKNOWLEDGEMENTS**

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**REFERENCES**

ARRAY SHAPE ESTIMATION USING MEASUREMENTS OF HEADING AND DEPTH SENSORS

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Abstract: Array shape of towed line array sonar usually deforms when the towed ship makes maneuver, and leads to performance degradation of detection and DOA estimation. One important way to solve this problem is to estimate the array shape real time and compensate the time delay caused by array deformation. In this paper, a method based on Ablow equation using measurements of heading sensors has been proposed to estimate the array shape by two steps: firstly, the array shape is predicted by the conventional Ablow equation by numerical approach; secondly, the measurements of heading/depth sensors have been used to check and modify the accuracy of array shape estimation. Simulation and sea trial results confirm the efficiency of this method.

Keywords: Towed linear array, Array shape estimation, .

1 INTRODUCTION

Towed array sonar is very useful for detecting noisy underwater target and has been popularly used since 1980s. One of the big deficiencies of this type of sonar is that array shape will be easily deformed when the platform makes maveour and causes degradation of beamformer for detection and bearing estimate. Various methods have been developed to cope with the problem of sensor-position uncertainty in towed linear arrays and the most common method is to place heading and depth-sensors at several points along the array to give localized horizontal/vertical information of the transverse displacements of the array. The shape of the linear array can be estimated by polynomial fitting to an acceptable accuracy if the number of depth/heading sensors is enough, but this idea is not feasible.
When a flexible array towed through water, the motion induced at the tow point, termed the “towed point induced” (TPI) motion, which propagate down the array can be governed the Paidoussis [1] equation. This equation forms the basis for most array shape-estimation methods in literature. Gray et al. [2] proposed a procedure for array shape estimation based on the Kalman filter algorithm. They discretize the Paidoussis equation in space and time so as to form a state-space representation for the transverse displacements at the array segments. Sensor measurements are related to the state variables and the Kalman filter is used to update the estimates over time. Feng et al. [3] proposed an approach to encounter the difficulties in coordinating both the sensor measurements and dynamics constraints, identifies the sources of discrepancy between sensor measurements and dynamics constraints as temporal distortion, make compensation for the distortion to improve the accuracy and consistency of array shape estimates.

We will study a time-domain approach based on the model provided by Ablow [4]. Under this literature, the array shape of a towed linear array can be numerically calculated by limited difference method. By using stable state solution and the-sea trial sensor data, some important parameters such as the drag coefficients can be inferred, and follow this way, several sophisticated approach such as Kalman filter can also be used for array shape estimation, which will form the future work.

2 THEORETICALLY MODELING

2.1 Governed Equations

According to Ablow[4], the shape of towed array can be separated into cable, linear array and tail rope, and the dynamics of these three parts can be depicted by several parameters such as tension, velocity, and posture angles. All these parameters can be demonstrated both in local coordinates \((t,n,b)\) and \((i,j,k)\), as shown in Fig.1.

These two coordinate systems can be related by the famous Euler angles \((\gamma, \alpha, \psi)\), where \(\psi\), \(\alpha\), \(\gamma\) denote the yawn, heading and pitch angles, respectively. If \(k\) is kept paralleled to the plane (shown in right part in Fig.1), then we have the following relation

\[
(t \ n \ b) = (i \ j \ k) D
\]

Where, transition matrix \(D\) can be given by

\[
D = \begin{bmatrix}
\cos \alpha \cos \gamma & -\cos \alpha \sin \gamma & \sin \alpha \\
\sin \alpha \cos \gamma & -\sin \alpha \sin \gamma & -\cos \alpha \\
\sin \gamma & \cos \gamma & 0
\end{bmatrix}
\]

Define the state vector of towed array, \(y = [T, v_1, v_2, v_3, \alpha, \gamma]^T\), and according to Albowz, the state of towed array should satisfy the following equation
\[ M\dot{y'} = N\dot{y} + q \]

Where, \( y' = \frac{\partial y}{\partial s}, \frac{\partial y}{\partial t} \). Coefficient Matrix \( M, N \) and vector \( q \) are given by

\[
M = \begin{bmatrix}
1 & 0 & 0 & 0 & 0 & 0 \\
0 & 1 & 0 & 0 & v_y \cos \gamma & -v_y \\
0 & 0 & 1 & 0 & -v_b \sin \gamma & v_b \\
0 & 0 & 0 & 1 & v_y \sin \gamma - v_b \cos \gamma & 0 \\
0 & 0 & 0 & 0 & -T \cos \gamma & 0 \\
0 & 0 & 0 & 0 & 0 & T
\end{bmatrix}
\]

\[
N = \begin{bmatrix}
-m \frac{e \frac{V_t}{1 + eT}}{m} & m & 0 & 0 & (mV_b - \rho AJ_b) \cos \gamma & -(mV_a - \rho AJ_a) \\
e & 0 & 0 & 0 & 0 & 0 \\
0 & 0 & 0 & 0 & 0 & 1 + eT \\
0 & 0 & 0 & 0 & - (1 + eT) \cos \gamma & 0 \\
-e \frac{(mV_b - \rho AJ_b)}{(1 + eT)} & 0 & m_i & (mV_a - \rho AJ_a) \sin \gamma - mV_t \cos \gamma & 0 \\
e \frac{(mV_a - \rho AJ_a)}{(1 + eT)} & 0 & m_i & -(mV_b - \rho AJ_b) \sin \gamma & mV_t
\end{bmatrix}
\]

\[
q = \begin{bmatrix}
W \sin \gamma + \frac{1}{2} \rho d \sqrt{1 + eT} \pi C_d U_t |U_t| \\
0 \\
0 \\
0 \\
\frac{1}{2} \rho d \sqrt{1 + eT} C_d U_b \sqrt{U_b^2 + U_a^2 - \rho AJ_a} \\
W \cos \gamma + \frac{1}{2} \rho d \sqrt{1 + eT} C_u U_a \sqrt{U_a^2 + U_b^2 - \rho AJ_a}
\end{bmatrix}
\]

2.2 Boundary Conditions

When towed through water, the linear array is connected to the towed platform by the towed cable and winch system, thus the velocity of the cable must equal to that of the platform. For this point of view, the towed array can be deemed as a “towed point induced” (TPI) motion. On the other hand, at the end of the towed array, i.e. the tip of tail rope, the tension caused by drag force of water is equal to zero. These characters of the towed array form the boundary conditions for solving equation (2).

1. Boundary conditions for towed point

As pointed above, the velocity at the towed point must satisfy the continuous relation, i.e.
\[
\begin{bmatrix}
0 & 1 & 0 & 0 & 0 & 0 \\
0 & 0 & 1 & 0 & 0 & 0 \\
0 & 0 & 0 & 1 & 0 & 0
\end{bmatrix}
\]

\[y(0,t) = \begin{bmatrix} v_x \ v_y \ v_z \end{bmatrix} D_T \]  \hspace{1cm} (6)

Where, \( v_x, v_y \) and \( v_z \) denotes the three components of the towed speed of the platform in the inertial coordinates; \( D_T \) is the transition matrix relates the local coordinates and inertial one.

2. Boundary conditions for tip of tail rope

According to the free end of tail rope, there are totally three conditions must be satisfied: one is the tension should be zero, and the other two are gradients of Euler angles equal to zeros. And the boundary conditions can be written as

\[ C(y(S,t)) + B(y(S,t)) y(S,t) + Q(y(S,t)) = 0 \]  \hspace{1cm} (7)

where

\[ C(y(S,t)) = \begin{bmatrix}
1 & 0 & 0 & 0 & 0 & 0 \\
0 & 0 & 0 & 0 & 0 & 0 \\
0 & 0 & 0 & 0 & 0 & 0
\end{bmatrix} \]  \hspace{1cm} (8)

\[ B(y(S,t)) = \begin{bmatrix}
0 & 0 & 0 & 0 & 0 & 0 \\
0 & 0 & m_i(\rho AJ_n - \rho AJ_b) \sin \gamma - m_i \cos \gamma & 0 \\
0 & 0 & m_i & 0 & - (m_i v_b - \rho AJ_b) \sin \gamma & m_i
\end{bmatrix} \]  \hspace{1cm} (9)

\[ Q(y(S,t)) = \begin{bmatrix}
0 \\
\frac{1}{2} \rho d \sqrt{1 + e T C_n \mu U_b^2 + U_b^2 - \rho A J_n} \\
W \cos \gamma + \frac{1}{2} \rho d \sqrt{1 + e T C_n \mu U_b^2 + U_b^2 - \rho A J_n}
\end{bmatrix} \]  \hspace{1cm} (10)

These two parts of boundary conditions and equation (2) form the conditions to determine the time-varying states of the towed array and can be solved via numerical methods.

3 NUMERICAL SIMULATIONS

As the drag coefficients of the cable, array and rope are quite different and the shape of towed linear array is very sensitive to these parameters, so should be estimated firstly. We use the sea-trial sensor data to make estimation of these parameters, as shown in Table 1. The parameters of towed linear array used in calculations are given in Table 2.
Table 1: Array depth in sea trial

<table>
<thead>
<tr>
<th>Segment</th>
<th>Diameter (m)</th>
<th>Mass/m (kg/m)</th>
<th>Length (m)</th>
<th>Young’s Modulus (kN)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cable</td>
<td>0.03</td>
<td>1.1</td>
<td>1500</td>
<td>$2.625\times10^7$</td>
</tr>
<tr>
<td>Linear array</td>
<td>0.07</td>
<td>4.04</td>
<td>250</td>
<td>$1.435\times10^7$</td>
</tr>
<tr>
<td>Tail rope</td>
<td>0.02</td>
<td>0.32</td>
<td>30</td>
<td>$2.743\times10^6$</td>
</tr>
</tbody>
</table>

Table 2: Parameters for simulation

From the sea-trial data, the drag coefficients can be inferred as: tangential drag coefficients ($C_t$) of cable/array/rope are 0.01492, 0.00897, and 0.02168, respectively; normal drag coefficient ($C_n$) of cable is 1.7687. Because the stable shape is independent on the normal drag coefficients of array and rope, these two parameters cannot be calculated by the data list in table1.

With the parameters acquired aforementioned, the shape of towed linear array sonar can be calculated by using equation Error! Reference source not found. 

Fig. 2 and Fig. 3 show the output of the heading and depth at the head and tail position of the 250m-long linear array when the length of towed cable is 1000 meters. From these two figures we can see that during the turn of platform, the heading of two points are quite different, it means that the array is not a straight line and the assumption of straight line shape for sonar signal processing is not proper. For lack of array element data, this degradation in array beamforming cannot be demonstrated here. Fig. 3 shows that the depth difference between these two points is not dramatic due to the neutrally buoyant and the depth at array end is slightly bigger than that of the head because of the density of the linear array is about 1.05g/cm$^3$, also slightly heavier than that of sea water.

4 CONCLUSIONS

We have studied the approaches for towed array shape-estimation problem using dynamic equation. The array shape can be calculated via numerical method by discretization of
temporal and spatial differential terms of array shape governed equations. In order to improve the estimate accuracy, some important parameters such as drag coefficients of the cable, array and rope are inferred by using the sea-trial data. In the future work, the output of heading/depth sensors will be fed to construct a new shape-estimator like Kalman filter to improve the real time estimate accuracy.

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RESULTS FROM OCEAN CURRENTS AND ACOUSTIC PROPAGATION MODELLING STUDIES IN SUPPORT OF THE INSTALLATION OF THE CTBTO HYDROACOUSTIC STATION HA04, CROZET ISLANDS, FRANCE

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\textbf{Abstract:} The Crozet Islands, the location in which CTBTO International Monitoring System Hydroacoustic Station HA04 is to be installed, is a particularly challenging oceanographic environment in the south-western Indian Ocean. A complex bathymetry, strong currents and internal tides make it challenging to find ocean bottom deployment sites for the hydrophone moorings that offer a good balance between hydrophone acoustic coverage and mooring sustainability. To fully understand these issues, the CTBTO initiated studies with the aim of developing a consolidated environmental and acoustic picture which could facilitate the identification of the most suitable locations for the hydrophone installations. These studies consisted of a review and reprocessing of available multi-beam data which provided the most complete bathymetric model of the area available to date, a 3-D time-domain ocean-current and internal tide modelling study which shed light on the complex interplay between circulation and internal tides in the area, and a 3-D parabolic equation basin-scale propagation modelling study of viable hydrophone mooring deployment locations in the presence of the complex environmental constraints. This paper introduces the studies briefly and shows some of the key results obtained.

\textbf{Keywords:} acoustic monitoring, propagation modelling, currents, oceanography
INTRODUCTION

Currently the CTBTO International Monitoring System (IMS) comprises 278 certified stations out of 321 planned stations and 16 laboratories, which provide the capability to detect nuclear tests in the atmosphere, underground or in the water. The four technologies which make this possible are: atmospheric infrasound (47 of 60 stations certified), seismic (147 of 170 stations certified), hydroacoustic (10 of 11 stations certified) and radionuclide (93 of 136 stations and laboratories certified). The focus of this paper is on the hydroacoustic component of the IMS, and in particular on HA04, the last and most challenging hydroacoustic station yet to be established.

Of the 11 hydroacoustic (HA) stations shown in Figure 1, 5 are T-stations, based on seismometers which pick up waterborne signals from events as they couple with the upslope crust in coastal areas, and 6 are hydrophone stations. HA03 in the Juan Fernandez archipelago (Chile), destroyed by a Tsunami in 2010, was recently re-installed in February/March 2014 (http://www.ctbto.org/press-centre/highlights/2014/welcome-back-ha03-robinson-crusoe-island/) and is currently undergoing testing and evaluation prior to revalidation. The project to establish HA04 on Ile de la Possession, Crozet Islands, French Southern and Antarctic Territories in the Indian Ocean, is currently under way.

Each hydrophone station (with the exception of HA01 Cape Leeuwin, Australia) consists of two triplets of hydrophones deployed North and South of an island to prevent bathymetric blockage. HA01 consists of one triplet, deployed to the West of Cape Leeuwin. The hydrophones are connected to the shore station by means of un-repeatered undersea electro-optical trunk cable. Each triplet is located between 30 km and 200 km from the shore. The hydrophone signals, which are digitized at the triplet and sent to shore via the trunk cable, are transmitted continuously in real-time to Vienna via a satellite link. Figure 2 shows a triplet with its trunk cable and the hydrophone moorings. The riser moorings are typically deployed at a distance of 2 km from each other and the hydrophones are floated in the ocean waveguide. In regions where the SOFAR channel is well established, this depth is as close to the channel axis as possible.
The hydrophone stations provide data in the band between 1 Hz and 100 Hz, with a sampling frequency of 250 Hz. The self-noise of the system is required to be 10 dB below ocean noise curves for typical low sea-state with low shipping, in order to maximize the potential detection range of the system. The maximum Sound Pressure Level (SPL) that the system must be able to process linearly without clipping is 185 dB re $\mu$Pa.

![Fig.2: Location of the Crozet archipelago and its two main islands (left) and schematic representation of a hydrophone triplet (right) connected to the shore by the trunk cable.]

The Crozet Islands are situated in a particularly challenging ocean environment, where major global ocean current systems (described in Sec. 3 below) interact to produce strong circulation and internal tide activity at all depths. Furthermore, the bathymetry around the islands is complex, characterized by canyons and steep slopes on the North side and a tectonic valley and a plateau with a shelf rich of volcanic cones to the South (Figure 3). Determining appropriate hydrophone locations therefore requires a careful consideration of the following environmental factors: a) the bathymetry, b) the currents (to prevent fatigue during the 20 year life-time of the system and to reduce noise induced by riser cable strumming and flow around the hydrophones), and c) the acoustic coverage which depends on hydrophone location as well as depth. The sound-speed profile in the area is mostly upward refracting.

This paper describes studies conducted in preparation of the establishment of HA04 which include a review of available bathymetric data and the production of a 100m resolution terrain model for the area [1], a study of the local ocean climatology and 3-D modelling of the circulation currents and internal tides in the area [2, 3], and acoustic coverage studies using a basin-scale 3-D propagation model [4]. These studies are briefly discussed below together with highlights of the results. In the final part of the paper, it is shown how data from the different studies are merged to provide a consolidated picture to guide engineering decisions regarding triplet location, hydrophone depth, riser length and cable routes.
BATHYMETRIC STUDY

Raw multi-beam data available at CTBTO from a 1998 survey at Crozet [5] and from a separate cable route resurvey conducted in 2003 were analysed by IFREMER and merged with high resolution data available from other surveys conducted by the French authorities [1]. An overview of the survey tracks used is given in Figure 4. The data along the tracks were collected by the RV Marion Dufresne II, which is operated by the Administration of the French Southern and Antarctic Territories and by the French Polar Institute Paul Emile Victor. The sonar used was a Thomson Seafalcon II multi-beam system, which operates at a centre frequency of 12 kHz, with transmit beam widths of 1.4° x 140° and receive beam widths of 3.6° x 2.4°[1]. The operating depth of the sonar ranges from 50 m to depths in excess of 5000 m. The multi-beam data were merged with global bathymetric databases, e.g. Gridone, to obtain the bathymetric terrain model used by the ocean currents and the acoustic modelling projects [2-4]. An example of such a merged bathymetry is shown in Figure 3.
OCEAN DATA RE-ANALYSIS AND MODELLING OF CURRENTS AND INTERNAL TIDES

The first step of the oceanographic study concerned the assimilation of data to be used to formulate boundary conditions and climatological forcing terms for the ocean models. Ocean climatology was obtained from the LEVITUS, World Ocean Atlas (WOA) 2005 and WOA 2009 databases. Large-scale ocean circulation was obtained by re-analysis with hydrodynamic simulations of assimilated data in the GLORYS2V1 re-analysis system. Besides providing boundary conditions for the high resolution ocean models around the Crozet Islands, the data assimilation and re-analysis provides useful information about the general climatology and dominating global currents in the area. The global circulation currents which dominate in the area are the Agulhas Return Current (ARC), the Sub-Antarctic Front (SAF) and the Polar Front (PF), which are shown in Figure 5 (top panel) at a depth of 1000m. The arrows in the figure represent the current vectors.

In the second step of the oceanographic study, the 2-D spectral finite-element tide model FES2012 [6] was used to obtain tidal diagnostics for the surroundings of the Crozet islands [2] (not presented here for reasons of space limitations). In addition to providing information about the general behaviour of the tides and the main components of the tidal

Fig.5: Top panel: large scale circulation yearly average from GLORYS2V1 re-analysis at 1000 m depth [2]. Bottom panels: local circulation averages at 50m and 1000m depths computed with the high resolution 3-D time-stepping model [3]. White areas in the top and bottom right figure indicate depth < 1000m, in the bottom left figure |v|>0.2m/s.
currents, the results were used to provide forcing conditions for the high resolution 3-D time-stepping finite difference ocean model Symphonie [7], which solved the relevant hydrodynamic equations on a 3-D mesh with a vertical discretization of 30 conformal depth layers, over a simulation period of one year (December 2008 to December 2009) with time increments of approximately 10 minutes [3]. The time-stepping model results for the local area around the Crozet islands show that the circulation currents interact with internal tides near the shelves to produce relatively strong currents throughout the entire water column near the islands (Figure 5, bottom panels). The situation is more dynamic on the South compared to the North side of Ile de la Possession because of a local counterclockwise deep circulation which is shielded by Ile de l’Est to the North [3].

Figure 6 shows results obtained from the time-stepping current model [3] for the one-year period at locations North (left panel) and South (right panel) depicted in the right panel of Figure 3. The mean and mean +/- standard deviation, as well as the maximum currents are stronger throughout the water column on the South compared to the North site, except for the top layers near to the surface. The intensification of currents between 400 m and 1000 m water depth on the South site is associated with internal tide activity, which is mostly absent at the North location shown here. A closer look at the histograms of the currents at the different depths shows that the distributions are mostly left skewed, with long tails to the right which imply that the maximum values are almost never reached (results not shown here for reasons of space). Comparison with moored current profiler spot measurements taken near the sea-floor at locations F1-F3 shown in Figure 3, which were conducted over 70 hour periods during the austral summer of 1998 [5], shows that the current model agrees well with the measured data in terms of current magnitudes, directionality and periodicity (results not shown here for reasons of space).

**LONG-RANGE 3-D ACOUSTIC PROPAGATION MODELLING**

To evaluate the acoustic coverage of the hydroacoustic station it is necessary to resort to a model capable of propagating sound to large distances (ocean basin scale and beyond) in range-dependent environments where energy is converted between different modes. The tool of choice for such problems is the parabolic equation model [8]. In addition to the vertical range dependence, the acoustic coverage of a hydroacoustic station depends also on lateral diffraction, such as shadowing around islands and other isolated bathymetric...
features as well as refraction from lateral gradients, as discussed for example in [9]. The 3-D parabolic equation (PE) model presented in [4,9] is based on the application of the split-step Padé PE approach to the along-range and cross-range directions, by first marching out all the bearings within a given range step, and then successively applying the PE propagation kernel horizontally to account for the lateral coupling. Sensitivity studies [4] have shown that 2 Padé terms in the azimuth and the commonly used 8 terms in the vertical are sufficient to yield converged results for the range of problems considered here. This makes it possible to solve the coupled 3-D problem with a computational effort of roughly 50% more than the typical Nx2-D parabolic equation modelling approach in which separate 2-D problems are computed for \(N\) independent bearing slices.

The left panel of Figure 7 shows the transmission loss (TL) map for Crozet, for the combination of one North and one South hydrophone. The most clearly visible 3-D propagation effects are the partial illumination of shadows behind islands and sea-mounts. The right panel of Figure 7 shows the acoustic coverage plotted as a function of the TL figure of merit (FOM), which is defined as the maximum TL that still enables a positive detection for a given source level, ambient noise and detection threshold. Displaying results in this format makes it possible to evaluate a variety of different combinations of source, noise and detection threshold parameters via one common representation of the TL model results. The relative increase in area coverage predicted by the 3-D model compared to the Nx2-D model can also be seen in the figure.

Fig. 7: Left panel, 3-D PE model transmission loss results for the combined acoustic coverage for hydrophones at 600 m depth, at location South-A and North-F (most western North location) shown in Fig. 3. View on the Indian Ocean basin. The frequency of the computation is 8Hz. Right panel: global area coverage as a function of the TL figure of merit (FOM) for a hydrophone at the South of Ile de la Possession.

In the first step of the search for a short-list of viable hydrophone locations, a total of 136 candidate locations were analysed with the acoustic model. By taking into account the results from the oceanographic study, and with a more detailed analysis of the bathymetry and slopes in the candidate hydrophone mooring deployment locations, it was possible to narrow down the search to 7 possible triplet deployment locations. Suitable hydrophone depths were determined by additional acoustic model runs for these 7 most promising locations. A geographic information system (GIS), which made it possible to present the different layers of information in one common picture, proved to be a helpful tool for tackling this complex problem, in which a multitude of parameters and results coming from different disciplines had to be considered in one common picture. Figure 8 shows an example of the GIS display.
Fig. 8: Bathymetry [1], hydrophone locations and cable routes displayed with a GIS tool. The white boxes show area coverage for different hydrophone depths (HD).

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Note: The views expressed herein are those of the authors and do not necessarily reflect the views of the CTBTO Preparatory Commission. Note regarding Crozet multi-beam bathymetry [1]: IFREMER product 2012 – Data collected by IPEV/IFREMER. No warranty granted by IPEV/IFREMER for any use of the results – Any rights reserved.

REFERENCES

USING CORRELATION MATRICES TO IDENTIFY TEMPORAL CHARACTERISTICS OF AMBIENT NOISE

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Abstract: In the deep ocean sound channel, biologic, anthropogenic and geophysical noise mechanisms combine to produce a very dynamic ambient noise field. As a complement to traditional spectral analysis techniques, correlation matrices can be used to characterize the ambient noise field by identifying ambient source mechanisms in frequency bands in which the acoustic levels tend to change together. Previous studies have investigated the frequency structure revealed by this approach. This study examines the use of correlation matrices as a method of identifying temporal characteristics of underwater ambient noise sources. The benefits and limitations of this technique will be demonstrated using ambient noise data recorded by the Comprehensive Nuclear-Test Ban Treaty Organization’s (CTBTO) hydroacoustic monitoring system.

Keywords: Ambient noise, CTBTO, correlation matrix
1. INTRODUCTION

In the deep ocean sound channel, sound sources from a very large area combine to produce a very dynamic ambient noise field. Furthermore, some noise types are already very dynamic in nature, due to constantly changing environmental conditions.

Correlation matrices are a useful tool for visually identifying different variables which tend to change together. While the correlation matrix has been effectively used in many fields, only a few studies have attempted to adapt it for use in underwater ambient noise analysis. Curtis et al. [1] have analysed about two years of ambient noise data from the Pacific Ocean using a correlation matrix, and identified a correlation between noise levels in the 200-500 Hz band and in the 12-15 Hz band. The correlation between noise levels in the two different bandwidths is the result of two different noise mechanisms which are similarly wind speed dependent. Similarly, Nichols and Bradley [2] have reported the correlation matrices of year-long recordings of ambient noise data from three different ocean basins. The correlation matrices were then used to identify the frequency ranges in which different source mechanisms were dominant.

This study examined the usefulness of the correlation matrix for the purpose of studying the dynamics of the ambient noise field under changing conditions. The examples discussed in this paper used correlation matrices to observe spectral differences between the noise fields during different surface wind conditions. First, the spectral differences between high and low wind speed conditions were examined. Next, noise produced by a single passing storm was monitored at three different stages of the storm. The primary data used were recorded by the Comprehensive Nuclear-Test Ban Treaty Organization’s (CTBTO) hydroacoustic station near Diego Garcia.

2. DATA

This study examined ambient noise data recorded during the year 2010 by two hydrophone arrays, moored near Diego Garcia and Wake Island. These stations are part of the hydroacoustic segment of the CTBTO International Monitoring System (IMS). A map of the two sites is included in Figure 1.

Each of the CTBTO hydroacoustic stations consists of six sound channel depth hydrophones, arranged into two triangular arrays, one to the northern side and one to the southern side of a host island. The hydrophones in the triangular arrays are spaced about 2 km apart, and the arrays are located between 20 and 190 km from the host island. The signal recorded at each of the moored hydrophones is digitized at a rate of 250 Hz and transmitted to a station at the host island by buried fiber optic cables. The signal is then sent by satellite to the CTBTO headquarters in Vienna for real-time monitoring [3]. A generic schematic of a CTBTO hydrophone station is seen in Figure 2. The Diego Garcia station has been actively recording since 2002, and the Wake Island station since 2007. While these hydrophone stations were built for the primary purpose of monitoring the world’s oceans for unsanctioned nuclear testing, the spatial and temporal extent of the data produced compose the most complete dataset available for the analysis of very low frequency underwater ambient noise.
In conjunction with the ambient noise recordings, this study used hourly wind speed measurements recorded at Diego Garcia and Wake Island. These datasets are maintained and distributed by the National Oceanic and Atmospheric Administration [4].

3. DATA PROCESSING

In order to build a correlation matrix from ambient noise data, raw data is first converted into a series of noise levels. For all the results shown in this paper, three minute long segments was selected from the data and converted to noise spectra using a 30-second FFT, with Hann windowing and a 50% window overlap. Depending on the time window of data to be analysed, different lengths of samples and FFT windows may be used. The first example used one three-minute sample per hour, whereas the second example used all 20 three-minute samples within each hour. The set of noise spectra can also be seen as a time series of noise level measurements at each frequency. To build a correlation matrix, the correlation coefficient is calculated between noise levels at each available pair of frequencies. The correlation coefficient in this paper is defined as [5]:

\[
r(f_1, f_2) = \frac{\sum_{i=1}^{n}(x_i(f_1) - \bar{x}(f_1))(x_i(f_2) - \bar{x}(f_2))}{\sqrt{\sum_{i=1}^{n}(x_i(f_1) - \bar{x}(f_1))^2} \sqrt{\sum_{i=1}^{n}(x_i(f_2) - \bar{x}(f_2))^2}},
\]  

(1)
where \( x(f_1) \) is the set of noise levels at frequency \( f_1 \). The calculated correlation coefficients form a diagonally symmetric matrix, where the i-by-j element represents the correlation coefficient between noise levels at \( f_i \) and \( f_j \). Since the diagonal elements represent the correlation between noise levels at the same frequency, the diagonal elements must always be exactly unity. The correlation matrix is then visualized in a color plot with logarithmic axes.

An example of a correlation matrix, calculated using a year of ambient noise near Diego Garcia, is seen in Figure 3. In this example, the visualized correlation matrix helps to identify frequency ranges of the ambient noise field in which noise levels tend to change together. For example, when the sound of an underwater earthquake reaches the hydrophone, the noise levels between 5 and 30 Hz all increase together, producing the region of high correlation in the middle of Figure 3, between 5 and 30 Hz. Similarly, the nonlinear effects of interacting wind-induced surface waves produce the bulk of the sound energy below 5 Hz. The spectrum of noise produced in this way is strongly dependent on the surface wind speed. In the correlation matrix, this noise mechanism appears as a high correlation region below 5 Hz, in the lower left corner of the image.

![Figure 3: Example of a correlation matrix, calculated from a year of ambient noise data near Diego Garcia (also seen in reference [2])](image)

4. EXAMPLE: EFFECT OF CHANGING WIND SPEEDS

   Below 5 Hz, the underwater ambient noise field is dominated by noise from nonlinear interactions of surface waves. The noise spectrum produced by this mechanism is directly dependent on the surface wave height spectrum, which is driven by the surface wind speed. According to the theoretical wave height spectrum proposed by Pierson and Moskowitz [6], the wave height spectrum increases with wind speed up to a saturation limit. Above this limit,
wave heights do not increase with wind speed, but follow a characteristic $f^{-5}$ spectrum. As wind speeds increase, the lower frequency limit of the saturation range decreases, so a wider range of frequencies in the wave height spectrum are saturated.

![Wind Speed Separated Spectra](image)

*Figure 4: Noise spectra measured during 2010 at Diego Garcia North, grouped by the surface wind speed at the time of measurement*

Using a model wave height spectrum, Hughes [7] and Lloyd [8] developed a theoretical spectrum for the underwater noise produced by nonlinearly interacting waves, which produce noise at twice the frequency of the active surface waves. Because of its direct relationship to the wave height spectrum, the noise spectrum also exhibits a saturation range, which has a characteristic spectral slope of $f^{-7}$. The expected saturation effect has been observed by McCreery et al. [9]. Figure 4 shows the ambient noise measurements from the north side of Diego Garcia during 2010, separated into spectra based on the surface wind speed. An $f^{-7}$ line is superimposed on the spectra to highlight the saturation effect seen in the CTBTO noise data.

Correlation matrices can be used to identify the low frequency limit of the saturation region. Figure 5a.) shows the low frequency (<10 Hz) corner of a correlation matrix, calculated from 381 three-minute noise spectrum samples randomly selected from the year 2010, while surface wind speeds were $\leq$2 knots. Similarly, Figure 5b.) shows the low frequency corner of a correlation matrix calculated from all 381 spectral measurements taken while the surface wind speeds were $\geq$15 knots. In both figures, the high correlation region in the bottom left corner is the result of wind-dependent noise.

During periods of low surface winds, spectral levels do not reach the saturation spectrum until about 4 Hz. As a result, when only noise spectrum observations from low wind speed conditions ($\leq$2 knots) are included, the spectral levels will tend to increase with wind speed all the way up to 4 Hz. Since the entire low frequency end of the spectrum tends to vary as a group while wind speeds vary, the correlation matrix shows a region of high noise level
correlation between frequencies below 4 Hz. When only high wind speed observations are included, the noise spectra are usually saturated above 2 Hz, so noise levels only respond to changes in wind speed below 2 Hz. Since variations in noise level between 2 and 4 Hz are essentially random, the noise level correlation is low between frequencies in the 2 to 4 Hz range. The only region of correlation produced by wind effects is below 2 Hz, because at high wind speeds, only these very low frequency signals are not saturated. The change in the upper limit of the wind produced correlation region can be used to identify the frequency limits of the saturation region of wind-induced noise.

**Figure 5: Correlation matrices calculated from ambient noise data recorded near Diego Garcia, for when surface wind speeds a.) were ≤2 knots and b.) were ≥15 knots**

5. **EXAMPLE: IMPACT OF A SINGLE STORM**

During a storm event, the wind speed at the surface exhibits rapid changes. Consequently, as the wind speeds shift, so do VLF noise levels. The plots in Figure 6 track the different stages of a storm recorded at Wake Island between January 16 and 25, 2010. Figure 6a.) shows the progression of wind speeds, with the three time periods of interest marked. Figures 6b.).-d.) show the correlation matrix computed from the noise during the periods of b.) quiescent winds before the storm, c.) increasing winds, and d.) decreasing winds.

During the period of relative calm before onset of the storm, the low frequency corner of the correlation matrix (Figure 6b.) does not show any effect of wind noise. Even though wind speeds during this period are mostly between 5 and 10 knots, which is strong enough to produce surface waves, the variation is not sufficient to generate a substantial enough variation in noise levels to be noticed in a correlation matrix. In the following two periods of the storm, when wind speeds first increased and then decreased, a strong correlation region was produced below 2 Hz (Figure 6c.-d.). Below 2 Hz, the noise levels influenced by surface winds were unsaturated, and therefore increase and decrease as wind speeds increase and decrease. This deterministic fluctuation in VLF noise levels produces the high correlation region seen in Figures 6c. and d.
Figure 6: a.) Hourly wind speed observations made near Wake Island during a storm, with three sections marked, corresponding to b.) the quiescent winds before the storm, c.) the increase in wind speeds and d.) the decreasing winds. Subplots b.), c.) and d.) show the correlation matrix computed from the three different segments of the data.

6. CONCLUSIONS

Correlation matrices are a useful tool for the analysis of underwater ambient noise. Particularly, they can be used to observe the effects of changing environmental conditions on the ambient noise spectrum. In an example shown in this paper, the correlation matrices were computed for two groups of noise spectra, grouped by two different surface wind speed categories. The correlation matrix which included data during low wind speed observations (≤2 knots) showed an area of high correlation below 4 Hz. The presence of this region demonstrates that the noise spectrum during low wind conditions does not reach saturation until around 4 Hz. The correlation matrix which, instead, only included observations made during high wind conditions (≥15 knots) only showed a strong correlation region below 2 Hz,
suggesting that the noise spectrum during 15 Hz winds is saturated above 2 Hz. A further example demonstrated the use of correlation matrices to track storm-induced noise at very low frequencies. In the example shown, wind speeds during the storm ranged from 5 to 25 knots. At these wind speeds, noise levels were unsaturated up to 2 Hz, and resulted in a strong correlation region below 2 Hz.

REFERENCES


ANTARCTIC’S SIREN CALL: THE SOUND OF ICEBERGS

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Abstract: While the steady increase in global shipping traffic has been identified as a primary cause of rising ocean noise level, in the Southern Hemisphere, the disintegration of large icebergs was found to be a significant noise source that influences the soundscape of ocean basins. Two large icebergs, B15a and C19a, calved off the Ross Ice Shelf in the early 2000s and drifted eastward to the warmer South Pacific Ocean in late 2007. For the next 1.5 years, while these icebergs were rapidly melting, they not only affected water circulation and the marine ecosystems in their vicinity but also influenced the low-frequency ambient noise level of the South Pacific basin. From 2008 to early 2009, the disintegration of B15a and C19a continuously projected loud, low-frequency sounds into the water column. The sounds propagated efficiently to lower latitudes, influencing the soundscape of the entire South Pacific basin. The icebergs’ sounds were recorded at Juan Fernández Islands (34°S, 79°W) by deep-water hydrophones maintained by the Comprehensive Nuclear-Test-Ban Treaty Organization (CTBTO). The sounds also propagated across the equator (~10,000 km from the icebergs) and were recorded by a deep-water hydrophone at 8°N, 110°W, maintained by the National Oceanic and Atmospheric Administration’s Pacific Marine Environmental Laboratory and Oregon State University. In the 30–36 Hz range, the noise level was ~6 dB and ~2.5 dB higher than in baseline years at the respective location. Spectrogram plotting shows that at CTBTO hydrophones, icebergs’ sounds dominate the frequency range below 100 Hz at which baleen whales vocalize. Some large icebergs in the Southern Ocean have lifespans over a decade. We suggest that icebergs calved off Antarctica can collectively generate a considerable amount of sound energy, which then propagates across ocean basins, influencing the ocean soundscape and marine environment for the duration of the icebergs’ disintegration.

Keywords: CTBTO, iceberg noise, polar acoustics, ambient noise

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INTRODUCTION

Marine animals rely on sound to aid in migration, feeding, and breeding [1][2][3]. Low-frequency ocean noise has been rising at an alarming rate (+3dB/decade [4]) and raising concerns as to its impact on marine ecosystems. There is a general consensus that anthropogenic sounds, most notably noise from ship and seismic air-guns, are the primary source of noise level increases [4]. These studies are based on historic data from US Navy hydrophones collected since the early 1960s [5],[6] from off the US West Coast near major shipping lanes. Long-term hydroacoustic records of the International Monitoring System (IMS) of the Comprehensive Nuclear-Test Ban Treaty Organization (CTBTO) have also become available for research purposes. Miksis-Olds et al. [7] found, from the records of Diego Garcia (HA08) in the Indian Ocean, that one of the arrays near the shipping lanes showed a significant increase in baseline noise levels caused by a recent increase in shipping, confirming that ship traffic is a significant contributor to ambient noise. Brown et al. [8] examined the entire CTBTO hydroacoustic data and suggested that the global average of low and high noise spectral curves agreed with Urick’s oceanic ambient noise curves [9] for moderate to heavy traffic shipping noise levels of 1984. Today’s light ship traffic noise between 10 and 100 Hz is at most ~7 dB higher than the Urick’s light ship traffic noise curves [9]. In contrast, the increase in sound levels in the 15–25 Hz is due primarily to increased baleen whale call energy [8]. However, it is still not clear how thousands of large ships scattered across a million square kilometres in mid-latitudes can collectively influence the soundscape across ocean basins and raise baseline noise levels, or if there is another natural or anthropogenic noise producing process that should also contribute to the global noise budget.

Since 1996, the National Oceanic and Atmospheric Administration (NOAA) and Oregon State University (OSU) have operated a deep-water autonomous hydrophone (AUH) array in the Southern Hemisphere, including the Bransfield Strait off the Antarctic Peninsula [10], Scotia Sea [11] and the eastern equatorial Pacific (EEP) [12]. In 2008, an EEP-NW record at 8°N, 110°W showed an unusually high level of wideband noise for a period of one year. Also, even higher elevated noise levels over the same time period, were observed at the IMS-HA03N hydrophone off Chile’s Juan Fernández Islands (33.4°S, 78.9°W) [13]. The source of the noise was identified as the two unusually large icebergs, C19a and B15a, which both drifted into the South Pacific. We discuss the timeline of the two icebergs’ disintegration, the nature of their wideband noises, and the northward propagation of sound across the equator in the Pacific Ocean. We also discuss the cryogenic iceberg breakdown processes in the open ocean, the seasonal fluctuations of iceberg volume as the major driving force of noise in the South Pacific, and how this process influences the soundscape across ocean basins.

ICEBERGS C19A AND B15A

From late 2007 to early 2008, two large tabular icebergs, C19a (~4,500 km² on 338, 2007) and B15a (~1,700 km²) drifted out to the sea-ice free regions of the open ocean off the Ross Sea Antarctic coast. These icebergs then drifted eastward in the South Pacific Ocean below 63°S. Most icebergs from the Ross Sea drift along the Antarctica coast westward driven by the Coriolis force; it is unusual for icebergs of this size to drift out far north and then move eastward in the South Pacific [13] and disintegrate there. In the South Pacific, the concentration of tabular icebergs is significantly lower than in the South Indian and South
Atlantic oceans [[14]]. It was, therefore, a rare event that two unusually large tabular icebergs appeared in the lower latitude of the Pacific Ocean. For the next 1.5 years, the two icebergs melted and broke apart continuously in the sea-ice free open water between 50°S and 60°S, losing volume by a process called “edge wasting” and “rapid disintegration” [15]. By early 2009, both icebergs had completely disintegrated to sizes too small to be tracked by satellite.

Figure 1a and 1b show a thinning process of B15a from 018, 2008 to 339, 2008. The surface area was reduced from ~1,700 km² to ~550 km² in 321 days, while keeping the original shape roughly the same. Figure 1c and 1d show the same thinning process of C19a. Figure 1b and 1c show edge wasting and rapid disintegration processes occurring at the same time.

Fig. 1. (a) Satellite picture of B15a on JD 018, 2008 at 61.4°S, 162.2°E near the Ross Sea area. Area estimate ~1,700 km². (b) B15a on 339, 2008 at 54.4°S, 157.2°W, ~2,700 km southwest from Christchurch, New Zealand. Area size ~550 km². (c) C19a on 338, 2007 at 63.6°S, 168.4°W. Area size ~4,500 km². (d) C19a on 364, 2008 at 58.8°S, 115.5°W. Area size ~1,200 km². Images are from Brigham Young University, Center for Remote Sensing.

C19a and B15a generated wideband acoustic noises (Fig. 2a) that were tracked by two hydroacoustic stations: one operated by the CTBTO since 2003 and located on the north side of Chile’s Juan Fernández Islands (yellow mark in Fig. 2b), and the other the NOAA/OSU EPP-NW (red mark in Fig. 2b). Wide station separation of ~5,600 km helped to improve the
array’s angular resolution. White triangles in Fig. 2b are the acoustic locations of iceberg sound sources (one per week) in 2008. Green and red lines are satellite tracks of B15a and C19a, respectively. Despite the long distances, ~10,000 km from sound sources to the receivers, the source location estimates matched reasonably well with the satellite tracks. The spectrogram and time series show a wideband and short-burst nature of the iceberg sound generated by edge wasting and rapid disintegration processes. The CTBTO and the NOAA/OSU hydrophone signals were sampled at 250 Hz and with 100-Hz anti-aliasing filter. Dziak et al. (2013) found bandwidth of another tabular iceberg’s (A53a) sound that exceeded 500 Hz [11], which was recorded in the Scotia Sea by a data logger with a 1-kHz sampling rate. The average source level of A53a was ~220 dB$_{rms}$ re 1µPa$^2$/Hz @ 1 m. It is expected that C19a and B15a’s source level is similar.

Fig. 2 (a) An 11-hour long acoustic record (top) and spectrogram (bottom) at EEP-NW on Julian day 169, 2008. A sequence of wideband iceberg noises arriving at the end of record. (b) Iceberg location estimates (white triangles) derived acoustically using the IMS-HA03N and EPP-NW arrays. Green and yellow lines are satellite tracks of B15a and C19a, respectively. Yellow and brown lines are the sea ice edge in the austral winter and summer, respectively. Blue lines are the great circle paths in September 2008 from iceberg C19a to IMS-HA03N (~4,500 km) and to EEP-NW (~9,600 km).
ICEBERG SOUND AFFECTING SOUNDCAPES ACROSS THE OCEAN BASIN

Figure 3a shows the contour of the monthly volume of ice distribution (in Gt) from small icebergs (<2 km in length) as a function of time and latitude (Hovmüller diagram) in the South Pacific region (150°E to 70°W) (reproduced from Tournadre et al. (2013) [16]). It shows that, in 2008, a large volume of ice departed the Antarctic coast, drifted northward, and reached ~50°S in early 2009. This increase of the volume of ice corresponds to C19a and B15a’s drift to the South Pacific region in late 2007 and subsequent disintegrations through 2008 and early 2009. It also shows that iceberg volume fluctuation is consistently austral-summer-high-winter-low. The black line is the sea-ice edge averaged over the South Pacific region, which also shows the same austral-summer-low-winter-high seasonality.

Fig. 3. (a) Logarithm of monthly volume of ice distribution (in Gt) of small icebergs in the Pacific as a function of time and latitude in the Pacific region. Black line is the sea-ice edge averaged over the Pacific. (b) Noise levels (in dB) relative to µPa²/Hz at EEP-NW (8°N, 110°W). (c) Noise levels (in dB) relative to µPa²/Hz at IMS-HA03N off the Juan Fernández Islands.
While C19a and B15a were breaking up in the sea-ice free open ocean, several thousands of kilometers away at the hydroacoustic station off Chile, the noise levels in 10–13 Hz (in red) and 30–36 Hz (in blue) rose by a maximum of ~12 dB and ~6 dB in September 2008, respectively, relative to the baseline of the previous two years (Fig. 3b). Similarly, the equatorial eastern Pacific noise levels in 10–13 Hz (red) and 30–36 Hz (blue) also went up by ~6 dB and ~2.5 dB, respectively (Fig. 3c). These two frequency ranges were chosen to avoid interference by the ubiquitous baleen whale calls in these areas [18]. The rise of noise levels at both sites corresponds to the period when large volumes of icebergs were moving out to the open waters at lower latitude.

ICEBERG DRIVING NOISE SEASONALITY ACROSS THE OCEAN BASIN

It appears that annual iceberg volume increase and decrease north of the sea-ice edge off Antarctica is influencing the seasonal variation in noise levels in the Southern Ocean, and appears to be extending across the equator up to 8°N. On average, approximately 400 Gt of icebergs are being transported annually to latitudes below 67°S and disintegrated annually [15] once exposed to by water temperatures above -1.8 °C and mechanical weathering forces due to current, wind, and waves [14]. At EEP-NW, the dynamic range of the annual average of noise levels showed clear seasonality, with austral summer highs and winter lows of ~4.5 dB and 3 dB in 10–13 Hz and 30–36 Hz, respectively. Austral summer high and winter low seasonality at EEP-NW matched with the seasonality of iceberg volume north of the sea-ice edge. At EEP-NW, wind speed has a seasonality of austral winter high and summer high [http://www.pmel.noaa.gov/tao/data_deliv/deliv.html], which is the exact opposite of the noise seasonality. Since there is no seasonality in shipping, and whale calls interference in these frequency ranges is unlikely, it seems apparent that the seasonal iceberg volume off Antarctica is a driving force for noise as far north as the equatorial Pacific.

At both hydrophone sites, the 10–13 Hz noise energy was stronger than the 30–36 Hz when iceberg sound was not blocked by a land mass. At IMS-HA03N, although iceberg noise from the Antarctic coast was blocked by the islands, there was still the same seasonality as the EPR-NW apparent in the 2003 and 2005 data. This could be a result of diffraction around the islands, or acoustic to seismic wave conversion through the island, coming through as acoustic wave again on the north side of the islands [19].

Both the EEP and Juan Fernández Islands hydrophones are near seismically and tectonically active areas. As a result, noise spikes are seen in Fig. 3b and 3c. Some spikes in early deployment of the EEP hydrophones from 1996 through 1998 can be associated with particular seismic and volcanic events [20]. A series of spikes can be seen in the IMS-HA03N noise levels in late 2009 to early 2010. The magnitude 8.8 Chile earthquake occurred on 27 February 2010, and the IMS-HA03N array was destroyed by the tsunami that followed.

CONCLUSIONS

In late 2007, large icebergs C19a and B15a drifted out from the Antarctic coast to the sea-ice free waters of the South Pacific, and their subsequent disintegration generated high-level acoustic noise that influenced the ambient ocean noise levels off Chile, as well the eastern equatorial Pacific across the equator, for 1.5 years. At the eastern equatorial Pacific, noise level rose by ~6 dB in 10–13 Hz and 2.5 dB in 30–36 Hz. The iceberg sound also raised the noise levels off the Juan Fernández Islands by 12 dB in 10–13 Hz and 6 dB in 30–36 Hz,
respectively. In the Southern Hemisphere, tabular icebergs off Antarctica are the major noise contributors. The surface sound generated by iceberg break-up process propagates efficiently to the north through surface duct, the Antarctic Convergence Zone, and deep-sound channel. Seasonal changes of iceberg volume north of the sea-ice edge appeared to be driving the noise level seasonality of austral-summer-high and winter-low in the entire Southern Ocean. Iceberg noise also reached across the ocean basin as far north as 8°N in the Pacific, influencing the seasonality with dynamic ranges of 4.5 dB and 3 dB in 10–13 Hz and 30–36 Hz ranges, respectively.

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REFERENCES


Session 11

Distributed Networked Systems for Surveillance

Organizers: Frank Ehlers and Arne Schulz
ADAPTIVE BAYESIAN BEHAVIORS FOR AUV SURVEILLANCE NETWORKS

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Abstract: Autonomous underwater vehicles (AUVs) present a low-cost alternative and/or supplement to existing underwater surveillance networks. The NATO Centre for Maritime Research and Experimentation (CMRE) is developing multi-sensor data fusion techniques and collaborative autonomous behaviours to improve the performance of networks of AUVs for the purpose of underwater surveillance. In this work, a range-dependent acoustic model for the predicted probability of target detection is combined with the detections observed on all available platforms in a Bayesian framework to compute a posterior distribution on target position for each ping. This posterior is then used to by the AUVs to collectively optimize their future actions based on a mission-driven measure of network performance. A Bernoulli random finite set (RFS) filter is used to jointly estimate both the target state and whether zero or one target is present. Simulation results are presented quantifying the performance increase using these adaptive behaviours over traditional pre-planned trajectories. Real-data results are also presented for adaptive behaviours running on two AUVs during the COLLAB13 trial conducted by the NATO Centre for Maritime Research and Experimentation (CMRE). Work supported by NATO Allied Command Transformation (ACT).

Keywords: Autonomous vehicles, surveillance networks, data fusion, detection, tracking,
1. INTRODUCTION

Autonomous underwater vehicles (AUVs) have many potential applications in underwater surveillance due to the fact that they are low cost, covert, persistent, unmanned, and capable of towing hydrophone arrays. While the sensor payload may be inferior to conventional assets, the performance gap may be filled by implementing intelligent autonomous behaviours which fuse the data from multiple platforms and navigate the individual AUVs to positions which optimize the performance of the entire network. The NATO Centre for Maritime Research and Experimentation (CMRE) is developing multi-sensor data fusion techniques and collaborative autonomous behaviours to improve the performance of networks of AUVs for the purpose of underwater surveillance. The problem of jointly optimizing the paths of a network of AUVs [1-4] falls under what is known as sensor management in the literature, which focuses on the sequencing and deployment of a heterogeneous network of sensors [5-6]. It is computationally expensive, and grows combinatorically with the number of vehicles in the network. While dynamic programming methods are capable of solving the optimization problem in certain cases, the problem quickly becomes computationally intractable as the size of the surveillance network increases. Furthermore, since underwater acoustic communications can be unreliable and have very low throughput, a decentralized fusion strategy is needed, allowing the vehicles to be able to act on incomplete information when messages from collaborating platforms or a command and control centre are unavailable. Given the limited computational resources available aboard a typical AUV, simplifications must be made to the global optimization problem to allow it to be computed in real time aboard each node in the network.

In this work, a range-dependent acoustic model for the predicted probability of target detection is combined with the detections (originating from both targets and clutter) observed on all available platforms in a Bayesian framework to compute a posterior distribution on target position at each ping. This posterior is then used by the AUVs to collectively optimize their future actions based on a mission-driven measure of network performance. A Bernoulli random finite set (RFS) filter [7-9] is used to jointly estimate both the target state and whether zero or one target is present. A derivation of the Bayesian model and acoustic model for the predicted probability of detection is presented in Section 2. The collaborative method of autonomously navigating a network of AUVs is described in Section 3. Simulation results then are presented quantifying the performance increase using these adaptive behaviours over traditional pre-planned trajectories. Real-data results are also presented for this adaptive behaviour running on CMRE AUVs during the COLLAB13 sea trial conducted by the CMRE in collaboration with the Italian navy.

2. BAYESIAN MULTI-VEHICLE TARGET TRACKING

This work considers a network of $N_A$ AUVs monitoring a surveillance region with the objective of estimating both the presence or absence of a target and the target state, if present. This is formalized as a Bernoulli random finite set [7-9] where the set $X_k$ has cardinality zero if a target is absent, $X_k = \{\emptyset\}$, and unity if present, $X_k = \{x_k\}$, where
is the target state with the position \( x_k^{(p)} = [x_k^i, y_k^i]^T \) and velocities \( \dot{x}_k^i, \dot{y}_k^i \). This work assumes a constant velocity model
\[
x_k = F x_{k-1} + v_k,
\]
where \( F \) is the state transition matrix. The term \( v_k \) accounts for acceleration and unmodelled dynamics.

### 2.1. DATA MODEL

At time step \( k \), in this work corresponding to an active sonar ping, a set of detections in bearing and time are observed by the \( s^{th} \) AUV, given by
\[
Z_{k,s} = \{z_{k,s,i}, \cdots, z_{k,s,m_s}\}, \tag{1}
\]
where \( m_{k,s} \) is the number of detections on sensor \( s \) at time \( k \). If the detections resulting from clutter are considered independent and identically distributed, the data distribution under the \( H_0 \) hypothesis can be written as
\[
f_{k,s}(Z \mid X_k = \{\emptyset\}) = m_{k,s}! \mu_c(m_{k,s}; \lambda) \prod_{i=1}^{m_{k,s}} f^{(c)}(z_i), \tag{2}
\]
where \( \mu_c(m_{k,s}; \lambda) \) is the probability mass function (pmf) of the number of clutter detections observed (see (1)), assumed to be Poisson with rate \( \lambda \), and \( f^{(c)}(z) \) is the probability density function (pdf) of the spatial clutter distribution, assumed to be uniform. Under \( H_1 \), the target is observed with a detection probability which is a function of source/receiver locations, receiver array parameters, and the geoacoustic parameters of the environment.

When a target is present, it is observed with a detection probability which in general depends on receiver array parameters, source parameters, target strength, and environmental parameters such as bottom scattering strength, water column sound speed, sediment density and sound speed, and sea surface roughness. Numerical and closed-form solutions exist of varying fidelity which can be used to calculate the predicted signal-to-noise ratio (SNR) which includes anisotropic reverberation. In this work, the SNR is calculated using the adiabatic normal mode program ARTEMIS, written at CMRE by Chris Harrison. If the envelope of the received acoustic data is assumed to be Gaussian under \( H_0 \) and \( H_1 \), the probability of detection can be computed for any false alarm rate.

The probability of detection \( P_D(x_k^{(p)}, u_{k,s}) \) can thus be calculated for a target at any point in the surveillance region, and for any possible future position of the receivers. This is a fundamental part of navigation strategies which optimize the future AUV positions, and will be discussed further in Section 3. Figures 4 and 5 give examples of the probability of target detection as a function of latitude and longitude for an area near Palmaria, Italy, the location of the COLLAB13 sea trial. If we assume that at most one target is present, all other contacts are clutter and are independent of the target’s state. The set density under \( H_1 \) is given by
\[
f_{k,s}(Z \mid X_k = \{x_k\}) = m! \left( 1 - P_D(x_k^{(p)}, u_{k,s}) \right) \mu_c(m; \lambda) \prod_{i=1}^{m} f^{(c)}(z_i)
+ (m-1)! P_D(x_k^{(p)}, u_{k,s}) \mu_c(m-1; \lambda) \sum_{j=1}^{m} f^{(c)}(z_j) \prod_{i \neq j} f^{(c)}(z_i), \tag{3}
\]
where \( f_{k,s}^{(t)}(z) \) is the target-originated pdf, see details in [3].

### 2.2 BAYESIAN TARGET STATE ESTIMATION

The target state can be optimally estimated from the posterior distribution, computed using Bayes’ rule

\[
p(X_k | Z_{1:k}) = \frac{f_k(Z_k | X_k)p(X_k | Z_{1:k-1})}{p(Z_k | Z_{1:k-1})}, \quad (6)
\]

where \( p(X_k | Z_{1:k-1}) \) is the prior distribution at time \( k \) and \( p(Z_k | Z_{1:k-1}) \) is a scaling factor. Since the sensors are conditionally independent given the target state, the likelihood function \( f_k(Z_k | X_k) \) can be written as the product of the likelihood functions at the individual sensors

\[
f_k(Z_k | X_k) = \prod_{s=1}^{N_s} f_{k,s}(Z_{k,s} | X_k), \quad (7)
\]

where \( f_{k,s}(Z_{k,s} | X_k) \) is the likelihood function of the \( s \)th sensor at time \( k \), see Eqs. (4) and (5). Target contacts are set apart from clutter contacts either by occurring where the prior pdf is high from a contact in the previous ping, or by occurring on multiple vehicles.

### 3. COLLABORATIVE AUV NAVIGATION

In this section, a method is described by which an AUV may plan its future position and heading based on the predicted probability of target detection and the data observed over previous pings. The objective is to choose the path of the AUV which maximizes some measure of performance, either for the individual AUV or in collaboration with other nearby platforms. It is assumed that all AUVs in the surveillance network are able to share their position \( u_{k,s}, h_{k,s}, \) and their detections, \( Z_{k,s} \). The full Bayesian posterior may then be calculated locally aboard each of the vehicles. The AUV uses the posterior target distribution and the performance prediction model which gives the probability of detection as a function of target and receiver position to choose the path which places the vehicle in the best position over the look-ahead period \( L \). The look ahead time must be sufficiently long to avoid “myopic” solutions, and short enough to be computationally tractable. Further, the optimization problem must consider only points reachable by the vehicle within the given time. The possible position of an AUV at time \( k+1 \) can be written as a function of its position at time \( k \) and the speed and heading changes, \( \delta h_{k+1,s} \) and \( \delta h_{k+1,s} \),

\[
u_{k+1,s} = u_{k,s} + (v_{k,s} + \delta v_{k+1,s}) \begin{bmatrix} \sin(h_{k,s} + \delta h_{k+1,s}) \\ \cos(h_{k,s} + \delta h_{k+1,s}) \end{bmatrix} \quad (8)
\]

\[
h_{k+1,s} = h_{k,s} + \delta h_{k+1,s} \quad (9)
\]

The speed and heading changes are usually restricted by vehicle dynamics, and can be further restricted to limit the size of the search space in the optimization problem. The
globally optimal result can be found by considering every possible speed and heading change at each time interval, and grows in complexity with the look-ahead time $L$. The path $\hat{U}_{k,s}$ is selected as the set of future positions $[u_{k+1,s}, u_{k+2,s}, \ldots, u_{k+L,s}]$ which minimizes the expected value of the chosen cost function $c(u_{k,s}, h_{k,s}, x_k)$, which is in general a function of AUV position and heading and target state, summed over all pings in the look ahead time.

$$\hat{U}_{k,s} = \{u_{k,L+1,s} \mid \arg \min_{u_{k,k+L,s}, i=k+1} \sum_{i=k+1}^{k+L} E[c(u_{i,s}, h_{i,s}, x_i)]\} \quad (10)$$

$$= \{u_{k,L+1,s} \mid \arg \min_{u_{k,k+L,s}, i=k+1} \sum_{i=k+1}^{k+L} c(u_{i,s}, h_{i,s}, x_i)p(X_i|Z_{1:k})dx_i\} \quad (11)$$

The expectation is taken over the Bayesian posterior distribution computed at each ping, as explained in the above section. Note that while the optimal path is selected at every ping, the path is recomputed as more information becomes available (every ping). Therefore the path taken by the AUV may be considerably different than the optimal path selected at any ping.

### 3.1. SINGLE AUV PATH OPTIMIZATION

The chosen cost function should be related to the mission objective. In an undersea surveillance application, this is generally to detect, localize, and/or track a target. One possible cost function is to maximize the expected probability of detection. For a single AUV this can be written as

$$c(u^{(p)}, h, X) = \begin{cases} 1 - P_D(x^{(p)}, u^{(p)}, h) & \text{if } X = \{x\} , \\ 0 & \text{if } X = \varnothing. \end{cases} \quad (12)$$

### 3.2. COLLABORATIVE PATH OPTIMIZATION

For a network of AUVs, the optimal performance over the AUV network may not be achieved by each AUV optimizing its individual performance. Taking into account the probability of detection observed by the collaborating vehicles can potentially improve the performance of a network of vehicles. One heuristic collaborative cost function would be to minimize the probability that all vehicles miss the target,

$$c(u_1^{(p)}, \ldots, u_s^{(p)}, h_1, \ldots, h_s, X) = \begin{cases} \prod_{i=1}^{S}[1 - P_D(x^{(p)}, u_i^{(p)}, h_i)] & \text{if } X = \{x\} , \\ 0 & \text{if } X = \varnothing. \end{cases} \quad (13)$$

The cost function is in this case a function of the position of the collaborating vehicles. Note again that other cost functions are not only possible, but may improve performance based on the specific objectives of the network.
3.3. AUV IMPLEMENTATION

The cost function in (13) or (12) can be now computed by averaging the summands $c(X_n,u_n)$ under the posterior distribution $p(X_n|Z_{1:k})$. An exhaustive search for the optimal solution is typically not feasible. Several approximate solutions have been proposed, among them dynamic programming strategies. Aboard an AUV with limited computational resources, dynamic programming is computationally too expensive to be computed on-board in real time. For this reason we further constrain the optimization problem to a smaller search space. The optimization space of navigation paths is constrained to be composed of trajectories at a fixed speed of 1 m/s, based on the AUV platforms used at CMRE. CMRE’s Ocean Explorer AUVs must maintain approximately this speed while towing arrays. The heading changes are forced to be constant during the planning horizon. The optimization space is then composed of smooth arcs, which is itself desirable in AUVs towing horizontal line arrays; sharp manoeuvres may deform the shape of the array and lead to a loss of array gain and/or large contact localization errors. An example of the paths contained in the search space are shown in Fig. 1.

4. RESULTS

In this section simulation and real data results are presented to demonstrate the adaptive behaviour described above and to quantify the performance of the adaptive navigation against conventional navigation in which AUVs are forced to be in predefined places or to move along predefined paths. In the following results, a target is assumed present from the first ping.

4.1. SIMULATION
Simulations were based on the Co-operative Littoral ASW Behaviour 2013 (COLLAB13) experiment, conducted by CMRE in the Mediterranean Sea off the coast of Porto Venere, Italy from June 29 to July 07, 2013. The locations of the target and AUVs are shown in Fig. 2. Groucho was deployed in the racetrack shown in green and Harpo as shown in red. Each racetrack was approximately 2 km in length. The target strength (TS) was 0 dB, and the PFA was $10^{-6}$. With the probability computed given the source, target, and receiver position, a contact is generated on the target by drawing from the Gaussian distribution $N((x_t, y_t), S)$, where $S$ is the target localization error and is a function of the sensor positions, signal, and array parameters. An additional five clutter contacts are generated uniformly over the area, with an SNR corresponding to the model predictions for an object with 0 dB target strength. Because of the left-right ambiguity present on horizontal line arrays, ambiguous contacts are also generated for each of the target and clutter contacts. Figure 2 shows both the racetracks for Harpo (red dotted line) and Groucho (green dotted line) and the adaptive paths for different realisations of the data in solid lines. The target track was a random walk, with small variations from a straight line from the southeast corner to the northwest corner of the surveillance region. A single realization is shown in black. In general, the adaptive tracks both decrease the distance to the target and keep the target on the broadside beam with respect to the conventional tracks. Figure 2 also shows the increase in the probability of detection as a function of ping number realized by the adaptive paths, averaged over 500 realizations. These results show the potential benefit of the proposed navigation strategy in this particular geometry.

4.2. COLLAB13 DATA

The above behaviour was tested with real data during the COLLAB13 experiment. The target (in this case an echo repeater) was towed by the R/V Alliance in a larger racetrack shown in black. The area search behaviour was run on OEX Harpo on July 05, 2013. OEX Groucho was running a separate adaptive behaviour to minimise the localisation error on an existing target track. The maximum turn rate on Harpo was set to 0.15 deg/sec to further reduce the possible array curvature. Acoustic communications performance was good, and a large degree of collaboration was possible. Contacts were consistently obtained on both vehicles, and the AUV Harpo first sailed southwest to pass near the target and keep it on the broadside beam as it passed from northwest to southeast. Figure 3 shows the target pdf and probability of detection map for Harpo reconstructed from the vehicle logs and MOOS database created during the experiment. In Fig. 3, the location of Harpo is marked with a white circle and the contacts generated by Harpo are shown in smaller circles. The position of Groucho is received via acoustic message and marked with a white diamond, along with five contacts also received from Groucho shown by the smaller diamonds. Both vehicles have contacts on the target, and the posterior target pdf is large in the cells near the true target position and smaller near other contacts. The navigation decision places the areas of high PD in the broadside beams on the areas where the target pdf is large. As the target continued southeast, Harpo then made a sharp turn to the north to maintain a good probability of detection on the target.

5. CONCLUSION

A method of navigation a network of AUVs has been presented based on the Bayesian posterior given all the available knowledge contained in the observations.
Figure 2. Simulations showing 10 realizations of the tracks of the AUVs Harpo (red) and Groucho (green) after 140 pings (left). The conventional racetracks are shown in dotted lines. On the right, the average probability of detection is shown vs. ping number for both adaptive and conventional navigation.

Figure 3. Results from Harpo during the COLLAB13 experiment. Harpo navigates closer to the target and puts the target on broadside with the highest probability of detection (upper panels). Harpo then makes a sharp turn to starboard to prevent the target from entering its endfire beams with low probability of detection (lower panels). The posterior pdf and probability of detection are shown in the left and right panes respectively.
collected by the network up to the current time. A range-dependent acoustic model is used to predict the target detection probability for all target and vehicle positions, and the future vehicle positions are selected which minimize a chosen cost function. A simple though likely suboptimal cost function is presented which improves the performance of a network of two AUVs compared to conventional pre-planned trajectories. Results were presented which quantified the improvement in average probability of detection and the localization root mean square error. Future work will focus on mission- and information-driven cost functions which are optimal for specific underwater surveillance missions.

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PARAMETER ESTIMATION FOR NON-COOPERATIVE MULTISTATIC SONAR

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\textbf{Abstract:} The main task of multistatic active sonar is the localization and tracking of objects of interest (targets). Therefore, a precise knowledge of the position of the acoustic sources as well as of its own sonar sensor position and heading is mandatory for each receiver in the multistatic active sonar system. In this paper we consider a receiver consisting of an antenna array, which is towed by an autonomous underwater vehicle (AUV). If the AUV is exploiting non-cooperative sources neither the position of the source nor the time of a transmission is known in advance. Only rough estimates can be gathered at the signal processing stage. In this case it is necessary to improve the available knowledge over time. We assume here that the positions of some targets are known according to a given uncertainty. These targets can be fixed objects like wrecks or small islands. In comparison to solely processing the direct blast, additional information is obtained by evaluating the reflections from known targets. In this paper we discuss the potential of using these reflections for improving the estimates of the system parameters, in particular the positions of the acoustic sources and the times of transmission. We present an algorithm for automatic parameter estimation, which is based on the multihypothesis tracking (MHT) filter, and discuss results for simulated data.

\textbf{Keywords:} Multistatic Sonar, Multihypothesis tracking, Parameter estimation
MOTIVATION

Multistatic sonar tracking based on stationary receivers aims for the surveillance of a given surveillance area and has been studied in detail e.g. by the members of the ISIF Multistatic Working Group (MSTWG) [1]. In cooperation with the Centre for Maritime Research and Experimentation (CMRE) in La Spezia (Italy) a multi hypothesis tracker (MHT) for multistatic active sonar was developed [2].

Some years ago, CMRE started to use Autonomous Underwater Vehicles (AUVs) cooperating in a network for surveillance operations [3] [4] [5]. Each AUV has to localize targets (submarines) robustly and precisely by evaluating the target's sonar echoes. Avoiding detection, i.e. a covert operation, requires a minimization of own emissions, like communication or navigation signals. For target localization and tracking, this is realized by a multistatic sonar configuration: The AUV is receiving the signals transmitted by spatially distributed acoustic sources.

A suitable sonar sensor consists of a linear array of hydrophones, which is usually towed by the AUV. While the AUV state can be determined relatively well, there is only poor knowledge of the array position and, especially important for multistatic tracking, of the array heading because it lacks of sensors of sufficient accuracy. An extension of the algorithm [2] was presented [6], [7], which is able to increase tracking accuracy by estimating these inaccuracies. This is realized by exploiting knowledge on the operation area by utilization of sonar echoes of stationary objects ("clutter", "fixed points") with known positions, see Fig. 1. We found that by the exploitation of multistatic sonar measurements we can even aid navigation of the AUV, if navigational data alone is not sufficient.

In our previous work we assumed that the receiver and the sources operate cooperatively, which means that the location of the source, the signal type and the time of the transmission are known (up to some uncertainty) at the receiver. If an AUV is exploiting foreign sources, neither the position nor the time of transmission is known in advance and only rough estimates can be gathered at the signal processing stage.

In this paper the capabilities of source parameter estimation (in position and time of transmission) using multistatic sonar measurements are discussed. In this case we assume an advanced navigational unit to be on board of the AUV, such that precise knowledge of the receiver parameters is available.

For the localization of the source position the bearing measurement of the direct blast can be exploited. If multiple receivers are available, the source position can be estimated

![Fig. 1: Utilization of sonar reflections from fixed points for parameter estimation](image1)

![Fig. 2: Bistatic geometry: Sound from the source at s is reflected by the target at x and received at o. θ is the heading of the receiver relative to north](image2)
from bearing-only measurements of the direct blast [8]. For a single receiver the localization problem can be solved by target motion analysis (TMA), which is also used in passive sonar applications [9]. This is known as the bearing-only tracking problem. A property of TMA is that the achievable estimation performance depends not only on the bearing accuracy, but also on the motion of the receiver and the source. For a stationary receiver, the position of the source is not observable. Additional measurements of time-delays given by multi-path reflections can provide improved estimation performance [10]. We follow a similar idea by utilizing reflections from known stationary objects. Since the active sonar source has a high source level, reflections from objects that are geographically separated can be obtained. A diversity of the geometry is generally advantageous in the estimation context. However, the availability of a high amount of reflections increases the need for appropriate data association strategies in order to identify the measurements of fixed points and discriminate false alarms and the measurements of moving unknown targets. This will be done here in the framework of Multihypothesis Tracking (MHT) [11].

The paper has the following structure: In the next section the measurement model is described. After that, we formulate the estimation problem in the Bayesian context and introduce a solution strategy based on the MHT. Results will be discussed for simulated data and different variants of the derived algorithm. Furthermore, the application of the algorithm to automatic path planning is outlined.

MEASUREMENT MODEL

The bistatic measurement depends on the target, source and receiver state vectors as illustrated in Fig. 2. The measurement equation with respect to a moving target for the azimuth angle $\varphi$ and time of arrival (ToA) $\tau$ and Doppler frequency $d$ is given by

$$
\varphi = \text{atan} \left( \frac{x-x_o}{y-y_o} \right) - \vartheta, \quad \tau = \frac{||x-s|| + ||x-o|| - ||s-o||}{c_s} + \tau_o, \\
d = -\frac{f}{c_s} \cdot \left( \frac{(x-s)^T}{||x-s||} \cdot (\dot{x} - \dot{s}) + \frac{(x-o)^T}{||x-o||} \cdot (\dot{x} - \dot{o}) \right).
$$

(1)

where $||.||$ denotes the Euclidian norm and $\dot{x}$, $\dot{s}$ and $\dot{o}$ denotes the velocity components of the target, source and receiver. The speed of sound is given by $c_s$.

The ToA is given relative to the first signal that arrives at the receiver. This typically corresponds to the direct blast (direct signal from the transmitter) and depends directly on the distance between the source and the receiver $||s-o||$. In order to capture inaccuracies in the measurement of the direct blast, we introduce the parameter $\tau_o$, which is described by the time difference of arrival of the direct blast and the first available signal.

The measurement equation of the direct blast is given by:

$$
\varphi^{DB} = \text{atan} \left( \frac{x_x-x_o}{x_y-y_o} \right) - \vartheta, \quad \tau^{DB} = \tau_o, \quad d = -\frac{f}{c_s} \cdot \frac{(s-o)^T}{||s-o||} \cdot (\dot{s} - \dot{o}).
$$

(2)

Grouping parameters according to receiver and system parameters $O = (o, \dot{o}, \vartheta, c_s)$, transmitter parameters $S = (s, \dot{s})$ and target parameters $X = (x, \dot{x})$ allows us to summarize the measurement equation by

$$(\varphi, \tau, d) = h(X, O, S) \text{ or } (\varphi^{DB}, \tau^{DB}, d^{DB}) = h^{DB}(X, O, S).$$

(3)
Generally, the measurements are not precise and are typically modeled by additive white Gaussian noise
\[ Z = h(X, O, S) + W \] and \[ Z^{BB} = h^{BB}(X, O, S) + W, \] (4)
where \( W \sim \mathcal{N}(0, R) \) has a Gaussian distribution with measurement covariance \( R \). The individual measurements \((\varphi, \tau, d)\) are assumed to be independent, such that \( R \) has diagonal form.

The measurement errors are typically caused by sensor inaccuracies, random fluctuations and modeling imprecisions. Additional errors are introduced by imprecisions in the estimates of the sensor parameters, for example due to the movement of the receiving platform. To capture these inaccuracies in the sensor model the prior knowledge of the sensor parameters is modeled by Gaussian random variables, i.e.
\[ X = \mathcal{N}(0, P_X), \quad O = \mathcal{N}(0, P_O), \quad S = \mathcal{N}(0, P_S). \] (5)

Thus, instead of modelling the parameter inaccuracies implicitly by an increase of the assumed measurement error, we introduce parameter state vectors in the estimation framework. This enables us to incorporate the sensor parameter inaccuracies explicitly in the tracking filter and even to improve the available knowledge by the application of estimation algorithms.

**BAYESIAN ESTIMATION**

In the following we assume the existence of \( n_k \) fixed targets \( X = \{x_1, x_2, ..., x_{n_k}\} \). Measurements of these known targets can be understood as realizations of the measurement random variable defined above. However, measurements can also be originated from unknown targets or false alarms. Thus, let \( Z_k = \{z^k_1, z^k_2, ..., z^k_{n_k}\} \) denote the collection of \( n_k \) measurements obtained at time \( t_k \) and \( Z_{1:k} = \{Z_1, Z_2, ..., Z_k\} \) the collection of all measurements up to time \( t_k \).

Our task is the estimation of \( p(S, O|Z_{1:k}, X) \). According to the Bayes theorem this can be calculated by sequential updating according to
\[ p(S, O|Z_{1:k}, X) \propto p(Z_k|S, O, X) \cdot p(S, O|Z_{1:k-1}, X). \] (6)

By introducing the association variable \( \alpha \) it is obtained that
\[ p(Z_k|S, O, X) = \sum_{\alpha} p(z_k^{\alpha}|S, O, X). \] (7)

A realization of \( \alpha \) is a vector of dimension \( n_k + 1 \), where \( \alpha(\ell) = s \) means that the \( \ell \)th (fixed) target is associated with measurement \( z^k_s \). The \( 0 \)th target stands for the direct blast. Thus, it holds that
\[ p(z_k^{\alpha}|S, O, X) = p(z_k^{\alpha(0)}|S, O) \prod_{\ell, \alpha(\ell) > 0} p(z_k^{\alpha(\ell)}|S, O, x_\ell) \cdot f_c(z_k^C), \] (8)
where \( z_k^C = Z_k \backslash \{z_k^{\alpha(\ell)}\}_{\alpha(\ell) > 0} \) and \( f_c \) denotes the probability density with respect to false alarms.
If measurements of different fixed points and the direct blast are independent in the association context, we can separate the measurement set by gating to obtain disjoint subsets. In this case the estimation problem can be formulated analogously to the multi-sensor tracking problem, which means that the state vector is updated for different fixed points in a sequential order. Classical tracking filters like the MHT [11] can be applied.

IMPLEMENTATION OF THE MHT FOR PARAMETER ESTIMATION

Our implementation of the MHT follows the sequential update scheme described in the previous section. In the case that the association independency assumption is not fulfilled a simple variation is applied. Associations of the same measurement to multiple fixed points simultaneously are neglected.

The update step in the MHT according to the non-linear measurement equation (3) is implemented using the Unscented Kalman Filter (UKF) [12] and Extended Kalman Filter (EKF) [13]. We will compare results for both types of filters.

To manage the growing number of hypotheses, gating, hypotheses merging and pruning techniques are implemented as described in [11].

In case of a poor prior knowledge of the source position the two Kalman Filters showed an unstable behavior. To improve the initialization of the source position, the initial estimate is generated from the combination of the azimuth measurement of the direct blast $\hat{\phi}^{DB}$ and the ToA measurement $\tau$ of a single fixed point

$$s_x = \sin(\varphi + \vartheta) \cdot r + o_x \text{ and } s_y = \cos(\varphi + \vartheta) \cdot r + o_y, \text{ where}$$

$$r = \frac{-0.5(\tau - \tau_0)c_s \cdot ((\tau - \tau_0)c_s - 2||x - o||)}{\sin(\varphi^{DB} + \vartheta)(x - o_x) + \cos(\varphi^{DB} + \vartheta)(y - o_y) + (\tau - \tau_0)c_s - ||x - o||.}$$

In our application the other sensor parameters (like $o$ and $\tau_0$) are only known according to prior uncertainty. To incorporate these uncertainties in the initial estimate we use an approach based on the Unscented Transform according to [2].

SIMULATION RESULTS

Our simulation scenario follows the setup of a real data experiment (GLINT'10) which was conducted by CMRE in 2010 [5]. The geometry of two sources, one receiver and three fixed points is shown in Fig. 3. A two-hour scenario is simulated (where the receiver is once moving around the triangle). Measurements are generated for all three fixed points with $P_B = 1$ and a mean number of 40 uniformly distributed false alarms per time scan.
Assumed measurement accuracy standard deviation (STD):

<table>
<thead>
<tr>
<th>Azimuth</th>
<th>ToA</th>
<th>Doppler frequency</th>
</tr>
</thead>
<tbody>
<tr>
<td>$\sigma_\phi = 3^\circ$</td>
<td>$\sigma_T = 1/15 \text{ s or } \sigma_T = 1/50 \text{ s}$</td>
<td>$\sigma_d = 5 \text{ Hz}$</td>
</tr>
</tbody>
</table>

Assumed prior knowledge of scenario parameters STD:

<table>
<thead>
<tr>
<th>Receiver position</th>
<th>Receiver velocity</th>
<th>Receiver heading</th>
<th>Source position</th>
<th>Propagation speed of sound</th>
</tr>
</thead>
<tbody>
<tr>
<td>$\sigma_O = 3 \text{ m}$</td>
<td>$\sigma_\dot{O} = 3 \text{ m}$</td>
<td>$\sigma_\theta = 1^\circ$</td>
<td>$\sigma_S = 500 \text{ m or } \sigma_S = 2000 \text{ m}$</td>
<td>$\sigma_{c_S} = 20 \text{ m/s}$</td>
</tr>
</tbody>
</table>

Fig. 4: Results for Tx 1 using standard initialization and assuming known association, $\sigma_T = 1/15 \text{ s, } \sigma_S = 500 \text{ m (left) and } \sigma_S = 2000 \text{ m (right)}$

For 500 Monte Carlo Runs the performance of the EKF and UKF-version of the MHT is compared to the Cramér Rao Bound (CRB) [14] in Fig. 4-Fig. 6. The results are displayed by the Root Mean Squared Error (RMSE) of the source position and root trace of the CRB. Outlier detection is implemented according to the Grubbs test [15]. Outliers have been removed from the error statistics and the percentage of outliers is displayed below the RMSE.

Fig. 5: Results of full MHT for Tx 1(left) and Tx 2 (right), $\sigma_T = 1/15 \text{ s, } \sigma_S = 2000 \text{ m}$
In Fig. 4 results for transmitter Tx 1 are shown for different assumptions on the available prior knowledge in the source position. An initialization of the MHT with poor prior knowledge of the source position leads to a decrease in performance (even when assuming perfect data association).

Fig. 5 shows results for Tx 1 and Tx 2 and the full MHT using the initialization procedure according to equation (9). Even for poor prior knowledge $\sigma_5 = 2000$ m stable results are achieved for both versions of the MHT. Fig. 6 shows the same results for an improved accuracy of the ToA measurement. In this case slightly worse results are obtained, which is reasonable due to the stronger effect of the non-linearity.

APPLICATION TO AUTOMATIC PATH PLANNING

In bistatic active surveillance scenarios a good estimation of the position of the acoustic sources and the transmission time of pulses might become a prerequisite for the application of advanced target tracking algorithms. Hence, the prediction of the performance of the parameter estimation methods should be used as input to the formulation of an internal goal function driving the AUV and its bistatic receiver to positions that offer the best geometrical properties. Since the geometrical properties depend on the spatial distribution of fixed points, acoustic sources and the receiver, the path planning related to the internal goal function has to reflect the changes of the estimation quality over the course of the paths under consideration. We propose the application of an A* algorithm [16] with the heuristic of the shortest travel time and a maximization of the estimation quality as a goal function.

\[ \sigma = \frac{1}{50} \text{s}, \sigma_S = 2000 \text{ m} \]

Fig. 6: Results of full MHT for Tx 1 (left) and Tx 2 (right), $\sigma_T = 1/50$ s., $\sigma_S = 2000$ m

CONCLUSION

The main contribution of the work presented in this paper is the seamless implementation of system parameter estimation within the MHT filter design. The benefits of this implementation are the robustness of the MHT framework for measurements in the highly variable underwater acoustic channel and the possibility to link predicted state uncertainties with control or planning tasks. The latter is one (of many) prerequisite for the realization of autonomous operations. A general drawback of MHT filter implementations is the introduction of latency: In good measurement conditions, a single ping estimation of
system parameters might be possible which would be much faster than running the MHT filter with present smoothing capabilities. However, in the discussed application of bistatic active wide area surveillance the achievable robustness is deemed to be more valuable than a short latency for the parameter application. Besides algorithmic improvements to the chosen MHT implementation, a proper path planning for the receiving platform could help to minimize the latency introduced by the proposed MHT implementation.

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OPTIMAL AREA COVERAGE IN AUTONOMOUS SENSOR NETWORKS

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Abstract: Autonomous sensor networks (ASNs) are a special kind of wireless sensor networks (WSNs) that rely on collaborative in-network data processing instead of routing aggregated data to a fusion center. Hence, ASNs are particularly useful in hazardous environments or areas that are inaccessible to humans. Each ASN application starts with the problem of network formation. Instead of deploying the network in a predefined topology, one considers mobile sensor nodes that distribute themselves autonomously in a region of interest (ROI) while avoiding collisions. Ultimately, the network should reach a quiescent state in which blanket coverage of the ROI is attained. We consider two algorithms for autonomous network distribution with collision avoidance, namely, Virtual Forces (VF) and Extended Virtual Spring Mesh (EVSM). VF considers the motion of individual nodes as a result of repulsion forces from neighboring nodes as well as obstacles, and an attraction force to neighbors in order to ensure swarm coherence. In addition, we introduce an exposure force that pushes the nodes into blind spots such as the shadow regions of obstacles. EVSM considers the network as a mesh of virtual springs that exert either an attractive or repulsive force on the nodes. We improve the algorithm by adding a repulsion force for obstacle avoidance. Our simulations show that a combination of VF and EVSM yields the best area coverage over time in terms of speed and spread.

Keywords: autonomous sensor networks, optimal area coverage, adaptive network formation, adaptive topology optimization
1. INTRODUCTION

In wireless sensor networks (WSNs), routing the aggregated network data to a fusion center can create a communication bottleneck and single point of failure [1]. A new class of WSN called autonomous sensor network (ASN) solves this problem by completely decentralizing data processing. The sensor nodes collaboratively process measurement data within their neighborhood and disseminate the local results throughout the network so that, ultimately, a global decision is formed [1, 2, 3]. Ideally, some or all of the nodes are equipped with appropriate actuators, allowing action to be taken without human intervention. Hence, ASNs can be particularly useful in hazardous and inaccessible environments [4, 5].

The artificial intelligence of ASNs is often inspired by nature. In wildlife, we can observe groups of animals that develop a global pattern of behavior, i.e., swarm intelligence, which emerges from limited and localized interactions of primitive individuals [3]. The concept of self-organization, which manifests itself in these swarms, is particularly interesting for the development of ASNs that can adapt to sudden changes in topology or the environment such as node failure or broken communication links [3]. This kind of ASN can be employed in applications such as target tracking and pursuit, in which the network autonomously organizes its topology and plans its motion path.

In this work, we consider the monitoring of shipping traffic in a harbor basin with an ASN. To this end, sensor nodes are deployed in a corner of the basin. They autonomously distribute themselves throughout the region of interest (ROI) and form a network that provides optimal area coverage while avoiding obstacles. In [3], the authors developed an ASN that forms and adapts its topology with the help of attraction and repulsion forces. Based on their findings, we introduce an extended distribution algorithm called Virtual Forces (VF), which distributes the ASN in an unknown ROI with the aim of maximizing area coverage. In addition, collisions with obstacles or other sensors are avoided and nodes group around obstacles to guarantee coverage of their shadow regions. An alternative distribution approach can be found in [4, 8], referred to as Extended Virtual Spring Mesh (EVSM). EVSM models the network as a mesh of virtual springs, which provides coverage of an arbitrarily shaped ROI. We improve EVSM in terms of obstacle avoidance and compare it to VF in order to find the algorithm that yields optimal area coverage.

The paper is organized as follows. Section 2 describes the network and measurement model. In Sections 3 and 4, we present the two distribution algorithms VF and EVSM, respectively. Section 5 compares the performance of both algorithms in terms of area coverage over time in a simulated scenario.

2. NETWORK AND MEASUREMENT MODEL

We model the ASN as an undirected graph $G(V, E)$ of nodes $V$ and edges $E$, where Node $k$ can communicate with Node $l$ if and only if Node $l$ can communicate with Node $k$. Furthermore, we consider only connected graphs, in which each node is connected to at least one other node, i.e., no separate subgraphs exist. In general, only nodes within a certain communication radius $R_{\text{com}}$ are connected. We call them neighbors and define the neighborhood $N_k$ of Node $k$ as

$$N_k = \{ l \in \{1, \ldots, N\} \mid \|x_l - x_k\|_2 \leq R_{\text{com}} \}, \quad k = 1, \ldots, N, \quad (1)$$
where \( N \) is the number of nodes and \( \| \cdot \|_2 \) is the Euclidean norm. Vector \( \mathbf{x}_k = [\rho_{x,k}, \rho_{y,k}]^T \), \( k = 1, \ldots, N \), denotes the location of Node \( k \) in the xy-plane. Furthermore, each node is a member of its own neighborhood and the communication is assumed to be ideal. In order to automatically guarantee a connected network when the ROI is covered, \( R_{\text{com}} \geq 2 R_{\text{sen}} \) is usually chosen as \( R_{\text{com}} \), with the sensing radius \( R_{\text{sen}} \) [7]. Conversely, a node spacing of \( R_{\text{opt}} \leq 2 R_{\text{sen}} \) is optimal for achieving 1-coverage [8].

Each Node \( k \) measures the noisy distance \( d_{l,k} \) and bearing \( \theta_{l,k} \) to Node or Obstacle \( l \) according to

\[
\begin{align*}
\tilde{d}_{l,k} &= d_{l,k} + \eta_d \quad \text{with} \quad d_{l,k} = \| \mathbf{x}_l - \mathbf{x}_k \|_2, \quad l, k = 1, \ldots, N, \\
\tilde{\theta}_{l,k} &= \theta_{l,k} + \eta_\theta \quad \text{with} \quad \theta_{l,k} = \tan^{-1} \left( \frac{\rho_{y,l} - \rho_{y,k}}{\rho_{x,l} - \rho_{x,k}} \right), \quad l, k = 1, \ldots, N,
\end{align*}
\]

where \( \eta_d, \eta_\theta \sim N(0, \kappa d_{l,k}^2) \) are zero-mean independent Gaussian random variables and \( \kappa \) is a positive weighting constant. Using the noisy distance and bearing measurements, Node \( k \) can estimate the noisy location \( \mathbf{q}_{l,k} \) of Node or Obstacle \( l \) as

\[
\mathbf{q}_{l,k} = \mathbf{x}_k + \tilde{d}_{l,k} \begin{bmatrix} \cos \tilde{\theta}_{l,k} \\ \sin \tilde{\theta}_{l,k} \end{bmatrix}, \quad l, k = 1, \ldots, N.
\]

### 3. VIRTUAL FORCES (VF)

The Virtual Forces (VF) algorithm is a bio-inspired distribution approach, which mimics the behavior of an animal swarm [3]. Each member of the swarm is subject to repulsion and attraction forces, which maintain the balance between avoiding collisions with neighbors or obstacles and forming a coherent swarm. The distribution problem can be solved by minimizing a local potential function \( J(\mathbf{x}_k) \), which is given in [9, 10] as

\[
J(\mathbf{x}_k) = \sum_{l \in N_k} \left[ a(d_{l,k}) - J_r(d_{l,k}) \right], \quad \forall k = 1, \ldots, N,
\]

using gradient-based optimization. Note that the sum is taken over the open neighborhood of Node \( k \). In addition, the repulsion function \( J_r(d_{l,k}) \) is maximum when both nodes occupy the same location. Thus, it prevents collisions, but it can also cause an excessive network spread. Therefore, an attraction term \( J_a(d_{l,k}) \) is introduced as a counterforce. Since the attraction grows proportionally to the node distance, swarm coherence is enforced. Ultimately, every pair of neighboring nodes attains an *equilibrium distance*, at which both forces balance each other, and the network enters a quiescent state. To this end, the forces are chosen such that the equilibrium distance is equal to the optimal node spacing \( R_{\text{opt}} \).

Each Node \( k, k = 1, \ldots, N \), starts with an initial guess \( x_k(0) \) and iteratively updates its location vector over time according to the dynamical system [3]

\[
\begin{align*}
\mathbf{x}_k(i + 1) &= \mathbf{x}_k(i) + \Delta t \mathbf{v}_k(i + 1), \quad i = 0, 1, 2, \ldots, \\
\mathbf{v}_k(i + 1) &= -C_1 \frac{\partial}{\partial \mathbf{x}_k(i)} J(\mathbf{x}_k(i)), \quad i = 0, 1, 2, \ldots,
\end{align*}
\]

where \( \Delta t \) denotes the time step and \( \mathbf{v}_k(i + 1) \) is the velocity vector at time instant \( i + 1 \). In addition, \( F_a(k, i) \) and \( F_r(k, i) \) denote the attraction and repulsion forces, respectively, and \( C_1 = 1 \text{ s/kg} \) is an auxiliary constant. In line with [9, 10], \( F_a(k, i) \) and \( F_r(k, i) \) are chosen as
with the attraction and repulsion coefficients $C_a$ and $C_r$, respectively. Note that the denominator of (9) includes the minimum clearance radius $R_{\min}$ and physical radius $R_{\phy}$, which are assumed to be equal for all nodes. This ensures that each node is protected by an impenetrable clearance region.

In order to avoid obstacles, we modify (7) by adding a second repulsion force $F^\text{obs}_r(k, i)$, which is given by

$$F^\text{obs}_r(k, i) = - \sum_{o \in L_k} \frac{C^\text{obs}_r}{(d_{o,k} - R_{\min} - R^\text{obs}_{\phy})^2} \frac{x^\text{obs}_o(i) - x_k(i)}{d_{o,k}}, \quad k = 1, ..., N, \quad i = 0, 1, 2, ...$$

with

$${\mathcal{L}}_k = \{ o \in \{1, ..., N_{\text{obs}} \} | d_{o,k} \leq R_{\text{sen}} \}, \quad k = 1, ..., N,$$

where $C^\text{obs}_r$ is the repulsion coefficient for obstacles and $R^\text{obs}_{\phy}$ denotes the physical obstacle size. In addition, $x^\text{obs}_o(i)$ is the location vector of obstacle $o$, $o = 1, ..., N_{\text{obs}}$, at time instant $i$, $i = 0, 1, 2, ...$, and $N_{\text{obs}}$ denotes the total number of obstacles. Depending on its strength, the second repulsion force may cause insufficient coverage of the regions behind obstacles. In order to address this problem, we extend the velocity vector update by another force called exposure [11]. To this end, the sensing region of Node $k$ is divided into a grid of $M$ squares, each of which is described by the location vector $p_m$, $m = 1, ..., M$, of its center point. In order to improve upon area coverage, Node $k$, $k = 1, ..., N$, should move towards the grid point $m_0$ that is least likely to be exposed to the neighbors of Node $k$. By performing an exhaustive grid search at every time instant $i$, this location is found to be

$$m_0 = \arg \min_m \sum_{l \in {\mathcal{N}}_k} \frac{\alpha_{l,m}}{\|x_l(i) - p_m(i)\|^2}, \quad i = 0, 1, 2, ...$$

where $\alpha_{l,m}$, $0 \leq \alpha_{l,m} \leq 1$, is the energy distortion factor as defined in [11]. This is equal to choosing the location that is least likely to be contained in the joint sensing region of the neighbors of Node $k$. By adding $\alpha_{l,m}$ to the equation, locations in the shadow of obstacles are preferred since the corresponding sensing link might be distorted or broken. Hence, sensor nodes considering the exposure term tend to group around obstacles in order to illuminate the area in their shadow. With the location vector $p_{m_0}(i)$, we can write the exposure force $F_e(k, i)$ as

$$F_e(k, i) = C_2(p_{m_0}(i) - x_k(i)), \quad k = 1, ..., N, \quad i = 0, 1, 2, ...$$

where $C_2 = 1 \text{N/m}$ is an auxiliary coefficient. Adding (10) and (13) to (7), the final velocity vector update now reads as follows:

$$v_k(i + 1) = (1 - \beta)v_k(i) + \beta (\varepsilon F_e(k, i) + (1 - \varepsilon)\left[F_a(k, i) + F_r(k, i) + F^\text{obs}_r(k, i)\right]) C_v,$$

with
\[
\varepsilon = \mathbb{I} \left( \| F_d(k,i) + F_r(k,i) + F_r^{\text{obs}}(k,i) \|_2 \right) \lambda \mathcal{C}_v. \tag{15}
\]

Note that \(\beta, 0 \leq \beta \leq 1\), is a forgetting factor for smoothing the velocity vector update with respect to the previous time step. In addition, \(\lambda, 0 \leq \lambda \leq 1\), is a coefficient for weighting the influence of the exposure force on the overall network distribution. Furthermore, \(\mathbb{I}(\cdot)\) is an indicator function that takes the value 0, if its argument is 0, and 1 otherwise. Thus, it turns off the exposure force once all nodes are at the equilibrium distance from their neighbors so that the network can enter a quiescent state.

4. EXTENDED VIRTUAL SPRING MESH (EVSM)

The Extended Virtual Spring Mesh (EVSM) algorithm enforces a network with triangular mesh structure by choosing the neighborhood according to the acute-angle test [6]. Similar to VF, the network distribution of EVSM is controlled by virtual forces. The first virtual force, \(F_{so}(k,i)\), is called self-organizing force of Node \(k, k = 1, \ldots, N\), at time instant \(i, i = 0, 1, 2,\ldots\), and is given by [8]

\[
F_{so}(k,i) = \sum_{l \in X(k) \setminus k} K_s \left( d_{l,k} - L_0 \right) \frac{x_l(i) - x_k(i)}{d_{l,k}} - K_d \mathbf{v}_k(i). \tag{16}
\]

It is based on the concept of virtual springs and applies to all nodes in the network. Just as mechanical springs, virtual springs are defined by their natural spring length \(L_0\), a spring constant or stiffness \(K_s\), and a spring damping coefficient \(K_d\) [8]. Thus, \(F_{so}(k,i)\) is either attractive or repulsive, depending on the distance \(d_{l,k}\) to the neighbors of Node \(k\). Ultimately, the distance between all nodes will approach the natural spring length \(L_0\), which corresponds to the optimal node spacing \(R_{\text{opt}}\) in VF. In contrast to VF, attraction and repulsion are linear forces of the same order and, therefore, similarly strong. This model should implicitly avoid collisions between sensor nodes, although neither the physical radius \(R_{\text{phy}}\) nor the minimum clearance radius \(R_{\text{min}}\) are considered in (16).

The second virtual force is the expansion force \(F_{\text{exp}}(k,i)\). It applies only to the boundary nodes of the ASN and enables them to pull the network into unexplored territory as well as expand beyond concave regions [4]. It is given by

\[
F_{\text{exp}}(k,i) = \begin{cases} 
K_s \left[ (d_{\text{left},k} - L_0) + (d_{\text{right},k} - L_0) \right] \mathbf{u}_{\text{exp}} - K_d \mathbf{v}_k(i), & \text{if on boundary} \\
0, & \text{otherwise}
\end{cases} \tag{17}
\]

where \(d_{\text{left},k}\) and \(d_{\text{right},k}\) denote the distances to the left- and right-hand frontier neighbors of Node \(k\), respectively. Furthermore, \(\mathbf{u}_{\text{exp}}\) is the unit vector pointing along the bisector of the sweep angle of these neighbors [6]. The boundary nodes of an ASN can be found using the notion of localized Voronoi polygons [7].

Assuming a normalized node mass \(M_k = 1\) kg, the acceleration \(a_k\) of Node \(k\) is given by the sum of self-organizing and expansion forces as

\[
a_k(i + 1) = F_{so}(k,i) + F_{\text{exp}}(k,i), \tag{18}
\]

from which the velocity and location update vectors can be obtained. However, the resulting move may not be legitimate if the node is in the vicinity of obstacles. In the original
formulation of EVSM [6], collision avoidance with respect to obstacles is attained by simply setting at time instant \(i + 1\)
\[
\mathbf{a}_k(i + 1) = -\mathbf{a}_k(i + 1),
\]
if \(\exists o \in \{1, \ldots, N_{\text{obs}}\} \text{ such that } \|x_k(i + 1) - x^{\text{obs}}_o\| < R_{\text{min}}. \tag{19}\]

If the reverse move is not possible, Node \(k\) does not move at all at this time step. Our experiments have shown, however, that this rule can hinder network distribution as it causes nodes in the vicinity of obstacles to oscillate. A better way to avoid collisions with obstacles is the addition of a repulsion force
\[
F^{\text{obs,EVSM}}_r(k, i) = -C_2 \sum_{o \in \mathcal{L}_k} \left[ x^{\text{obs}}_o(i) - x_k(i) \right], \quad i = 0, 1, 2, \ldots, \tag{20}\]
with
\[
\mathcal{L}_k = \{ o \in \{1, \ldots, N_{\text{obs}}\} \mid d_{o,k} \leq R_{\text{sen}} \}, \quad k = 1, \ldots, N, \tag{21}\]
to (18). Consequently, the total node movement is dictated by the sum of all three forces.

5. SIMULATION RESULTS

We consider a square harbor basin of dimensions 50 m \(\times\) 50 m, which is bounded by a quay wall on the northern, western, and southern sides. It contains four obstacles with a radial dimension of \(R_{\text{phy}}^{\text{obs}} = 2\) m. They are placed at random uniformly outside the launching area, which is given by the 14 m \(\times\) 14 m square in the lower left-hand corner of the harbor basin as shown in Fig. 1(a). Here, a total of \(N = 30\) sensor nodes are deployed uniformly at random and sequentially in packets of size \(N_{\text{launch}} = 5\). A new packet is launched as soon as the previous one has left the launching area. Since the nodes are deployed closely together, a strong repulsion quickly forces them apart. As the nodes drift further away from each other, the repulsion between them declines and with it the speed at which the individual nodes move. Once the average velocity of the network is below a given threshold, which we have empirically found to be 50 \% of the maximum speed \(v_{\text{max}} = 1\), the network has spread so far that the launching area is likely to be empty and a new packet of nodes can be deployed.

We perform \(N_{\text{MC}} = 200\) Monte Carlo runs to evaluate the performance of VF and EVSM in terms of area coverage over time. The coverage ratio (CR) is a measure of how good a known ROI is covered at time instant \(i\). It is given by

![Fig. 1: Harbor surveillance: (a) Exemplary harbor basin with highlighted launching area. (b) Exemplary network with its corresponding boolean coverage map in (c).](image-url)
where $A_{\text{cov}}(i)$ denotes the coverage area of the ASN at time instant $i$ and $A_{\text{ROI}}$ is the area of the ROI. $A_{\text{cov}}(i)$ can be determined graphically by drawing the quantized boolean map of the union of the sensing regions of all nodes as well as the areas occupied by obstacles and comparing it to the ROI. An exemplary ASN as well as its corresponding boolean map are depicted in Fig. 1(b) and Fig. 1(c), respectively.

The CR of VF and EVSM is given by the blue and green curves in Fig. 2, respectively. EVSM is able to attain a higher CR than VF with a value of 0.82 compared to 0.79 after 500 time steps. An explanation for this can be found in the different repulsion forces of VF and EVSM that are compared in Fig. 3. Although the repulsion force of VF goes to infinity when a neighboring node or obstacle approaches the maximum clearance radius $R_{\min}$ of Node $k$, it quickly drops towards zero as soon as the distance between them grows again. As a result, the total repulsion force affecting the sensor nodes is not strong enough to push the distance between them to the optimum, $R_{\text{opt}}$. Hence, the total network spread of VF is smaller than that of EVSM, which also leads to a smaller CR as less area is covered.

Due to the faster network spreading, VF already reaches its saturation level of 0.79 after 180 time steps. Although coverage speed is not important in our simulations, it might be a critical factor in practical scenarios such as target tracking. In this case, an optimal area coverage should be attained as fast as possible, in order not to miss an important event that might occur before the ROI is fully covered. In terms of area coverage, the optimal distribution algorithm is, therefore, given by a sequential combination of VF and EVSM. During the launching phase, we resort to VF in order to spread the network as fast as possible. As soon as all nodes are deployed, we switch to EVSM, which optimizes the network topology and maximizes the area coverage. The performance of the combined algorithm VF+EVSM is given by the red curve in Fig. 2. Clearly, VF+EVSM exhibits the best coverage performance. Up to time step 100, i.e., during the launching phase, the CR is identical to VF. Subsequently, we switch to EVSM and the slope becomes less steep but instead of going into saturation, the CR keeps rising to reach a value of 0.87 at time step 500. The total CR is, therefore, even higher compared to EVSM. The combined algorithm could be further improved by moving the switching point into the beginning of the optimization phase. However, finding the optimal switching point is difficult, especially in a decentralized fashion.
6. CONCLUSION

We have presented two algorithms for autonomous network distribution with obstacle avoidance in ASNs. The performance of VF was compared to an improved version of EVSM in terms of coverage ratio. While VF exhibits a faster network distribution than EVSM, its coverage ratio quickly goes into saturation. Under EVSM, however, network distribution is slower but a higher coverage ratio can be attained. Ultimately, a combination of VF and EVSM was found to yield the best area coverage over time with respect to the covered area as well as the coverage speed. As a next step, we will investigate the optimal switching point between VF and EVSM as well as the effects of non-ideal communication.

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REFERENCES

ACOUSTIC COMMUNICATION AND LOCALIZATION IN AUV
COOPERATIVE SURVEYS

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Abstract: The experimental results in acoustic communication and localization obtained with the “Typhoon” Autonomous Underwater Vehicle (AUV) in the CommsNet13 field trial are presented. The “Typhoon”s are a set of three AUVs developed by the Authors within the framework of the “Thesaurus” project, funded by the Tuscany Region, aiming at developing techniques for systematic cooperative autonomous exploration of marine archaeological areas by AUVs. The CommsNet13 experiment, which took place in September 2013 in the La Spezia Gulf, North Tyrrhenian Sea, was organized and scientifically coordinated by the NATO S&T Org. Ctr. for Maritime Research and Experimentation (CMRE); it included among its objectives the evaluation of on-board acoustic Ultra-Short Base Line (USBL) systems for navigation and localization of AUVs. For the experimental conditions encountered, the results obtained show that, by integrating on-board navigation systems (Inertial Measurement Units) with acoustic fixes the navigation error can be limited and re-set. With our communication/localization system, designed for the team operation, latency was however a non-negligible factor.

Keywords: Autonomous Underwater Vehicles; Acoustic Communication; Distributed Sensor Networks
1. INTRODUCTION

The potential of Autonomous Underwater Vehicles (AUVs) working as a team in sampling, monitoring and surveillance has been realized since quite a long time [1]. However, the autonomous operation of a team of AUVs, with distributed decision capabilities, is still in the experimental research phase. The most relevant obstacles to the operational implementation of the concept reside in the limitations of the acoustic channel for inter-vehicle communications, and in the uncertainties in underwater navigation associated to the use of tactical grade Inertial Navigation Systems (INS). The latter is often used on board of AUVs in order to limit the vehicles cost; however, the typical horizontal navigation error drift from a tactical grade AUV can be of several thousand meters per hour [2]. Most AUVs employ Doppler Velocity Loggers (DVLs) as auxiliary sensors in order to increase the time window of reliability of dead reckoning navigation, but in the final end, even with DVLs, the navigation error drift requires a re-setting of the absolute position through re-surfacing and acquisition of a Global Positioning System (GPS) fix, at the very best after few (2-3) hours of underwater operation.

As for navigation accuracy, there are alternatives to the resurfacing and GPS fixing procedure: the first commercial AUV, the Hugin, used the HiPap Ultra Short Base Line (USBL) acoustic positioning system operated from an ancillary ship, and the acoustically estimated position was then communicated to the vehicle thanks to the acoustic modem capabilities of the HiPap transducers [3]. Another alternative is represented by the use of Long Base Line (LBL) acoustic positioning systems [4]. These acoustic positioning techniques have been designed for single-vehicle operations. However, whenever a team of AUVs is considered for search/surveillance/survey missions over wide areas, the use of auxiliary ship(s) is indeed a strong limitation for the team autonomy, and ultimately limits the cost efficiency of the AUVs operation. The use of a LBL system requires the a priori definition of the survey area, which may or may not be an option for the mission at hand.

Motivated by the task of archaeological relict search over extended areas, our research group has been developing techniques for acoustic communication and cooperative localization for AUV teams, based on the commercial availability of acoustic modems with USBL capabilities at affordable costs. The basic idea is to use a USBL-capable vehicle at the sea surface, i.e., with GPS availability, to localize and georeference the submerged members of the team. The georeferenced position is then communicated acoustically to the submerged vehicles, and exploited as an alternative to GPS fixes. Note that, in order to cooperate toward the mission goal, the AUVs need also to communicate among themselves information regarding the mission status [5], [6]. An architecture for concurrent cooperative communication and localization has been presented by our group [7], [8] in the framework of the “Thesaurus” project [9]. In any system considering concurrent communication and localization, it has to be carefully evaluated the trade-off between the two concurrent tasks: in particular, communication in the team is achievable through some networking structure; the network structure in turn implies an overhead that results in communication delays (in addition to those traditionally experienced and due the physical characteristics of the underwater acoustic channel). Are the acoustic fixes communicated at a fast enough rate to avoid unacceptable drifts in the AUVs navigation? In this paper we report some experimental results to help to evaluate the performance trade-off at least in the case of our own developed communication/localization system. The data have been gathered within the CommsNet 13 cruise, organized by the NATO.
Centre for Maritime Research and Experimentation (CMRE) in September 2013, and taking place in the waters of the Gulf of La Spezia, Italy, North Tyrrhenian Sea.

The paper is organized as follows: in the next section the communication/localization scheme proposed in [7], [8] is briefly reviewed, for self-consistency, but omitting the implementation and algorithmic details that have already been reported in the cited works. In Section 3 the experiment set-up is described. In Section 4 the localization and underwater navigation results are presented. Finally, conclusions are given.

2. CONCURRENT ACOUSTIC COMMUNICATION AND LOCALIZATION

The communication/localization algorithms of the “Thesaurus” project [7], [8] have been implemented on the project-designed vehicles, the “Typhoon” AUVs, Fig. 1. Three Typhoons have been built within Thesaurus, differing from each other in terms of payload instrumentation and communication equipment. In particular, as for the communication aspect which is of interest here, one vehicle is equipped with a USBL-capable modem, while the other two are equipped with standard modems [10].

The vehicles establish a communication network based on a time-sharing bi-directional/broadcast communication scheme. We have opted for a Time Division Multiple Access (TDMA) broadcast messaging scheme, with no multi-hop (hence routing) capability. In the implemented network architecture, the bottom layer is represented by acoustic modem/USBL, which manages the physical transmission of the signals. The modem also implements collision avoidance techniques (Medium Access Control - MAC) and provides basic network functionalities, including an addressing system that can be exploited at the link layer.

Fig. 1: two of the three Typhoon vehicles – the top one with standard acoustic modem and payloads for underwater search, the bottom one with USBL modem

Acoustic Modem

Side-scan Sonar

Lights, DVL, laser scanner

uw cameras

USBL Modem
The medium access control is completed through a time division mechanism: time is divided into windows and each node/vehicle is assigned a time slot where it has to concentrate all its communication burden. The on-board data from the INS (in the Typhoon case consisting in X-Sense Inertial Measurement Unit) are fused with all possible other sources of information, i.e., ground velocity from DVL (if available), GPS data (when on surface), acoustic ranges (determined from message exchanges), georeferenced position messages (as received from the USBL vehicle). The fusion algorithm is a version of the Extended Kalman Filter (EKF) that takes care of the difference in rate of the various information, and in particular that the information from the communication system, either range only or complete geolocalization, arrives asynchronously with respect to the other sensors, and also at irregular intervals, due to the delay in transmission and the disturbances in the channel, including occasion fading.

3. EXPERIMENTAL SET-UP

The CommsNet 13 experiment has been organized by CMRE with the main objective to test the performance of several acoustic communication and localization systems using underwater networks. Several teams from different institutions, each one interested in testing different systems, have been involved in the experimentations, held with the support of NRV Alliance. All the teams were using Evologics modems, to guarantee compatibility at the physical level. A complete description of the activities of the Thesaurus project team within CommsNet 13 cruise is being reported in [11]. Here we report on the data (communication and navigation) gathered with the USBL-equipped Typhoon (bottom of Fig. 1). The test with the Typhoon AUV consisted in:

- deploying some (5 in our case) modems at the sea bottom, in unknown location;
- with the USBL AUV at the sea surface (i.e., with GPS contact), gather position data from the bottom-moored modems, in order to georeference them;
- use the georeferenced bottom-moored modems to get acoustic fixes through communication with USBL modem;
- integrate the acoustic fixes with the on-board INS data (the USBL AUV does not have a DVL sensor).

The experiment is done with the USBL vehicle at the surface, in order to have GPS as ground truth data. Moreover, the EKF is initialized using GPS data as well, in order to avoid initial conditions problem. Note that the acoustic localization data are obtained using the TDMA communication protocol, so that every modem is interrogated in a round robin fashion. The localization system is a hybrid between classic LBL and USBL methods: indeed, the vehicle is using a USBL sensor over a net of LBL nodes.

The experiment took place within the La Spezia harbor, in very shallow water (4 to 16 m depth), with horizontal distance between the various communicating nodes always less than 1000m (Fig. 2). The measured sound speed profile at the time of the experiment was constant with depth (1533 m/s, water temperature 23 °C, salinity 38.08 psu) in the early morning, gradually developing a downward reflecting gradient during the day. Sea state was 0 in the morning, with a gentle breeze developing in the afternoon and leading to sea state 1 – 2. The modems (both standard and USBL) operate with approximately 8 KHz bandwidth, centered around 30 KHz [10].
Fig. 2: experimental configuration for the CommsNet13 exercise reported. The black dots indicate the seabed modem nominal positions; the yellow triangle the nominal route of the USBL Typhoon. Note that one of the bottom-moored modem is a USBL modem, used as a standard one.

4. RESULTS

Bottom-moored modems georeferentiation is reported first. In Fig. 3 the clusters of individual acoustic fix from each of the bottom-moored nodes are reported, for both the morning and the afternoon of the experiment day. Note that both in the morning and afternoon outliers are present from modem M1. In the afternoon, possibly because of the downward reflecting gradient, there was no communication with modem F2 and just one fix for F1. After outlier removal, the clustered acoustic fixes from each modem have been averaged, in order to get a mean georeferenced position, and the standard deviation computed and reported in Table 1. Note that the uncertainty in the georeferentiation is caused by the inherent uncertainty in the USBL fix, by the motion of the moored modems, and by the GPS error.

Fig. 3: estimated position (dots) and mean value (cross) of the bottom moored modems from the surface measurement of the USBL Typhoon, in the morning (left) and in the afternoon of the experiment day.
Table 1: standard deviation in bottom-moored modem absolute georeferentiation

<table>
<thead>
<tr>
<th>Bottom-moored modem identifier</th>
<th>Standard deviation of position estimate (m)</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>morning</td>
</tr>
<tr>
<td>M1</td>
<td>0.8</td>
</tr>
<tr>
<td>M2</td>
<td>4.1</td>
</tr>
<tr>
<td>USBL</td>
<td>2.0</td>
</tr>
<tr>
<td>F1</td>
<td>5.2</td>
</tr>
<tr>
<td>F2</td>
<td>4.5</td>
</tr>
</tbody>
</table>

The runs with the USBL Typhoon navigating at the surface, which are relevant for this paper, have taken place in the morning. In Fig. 4 the Typhoon repeated paths above the nominal trajectory are shown, together with the acoustic fixes from USBL and the GPS ground truth at the time of the acoustic fix reception. In Table 2 the numerical values of the discrepancy between acoustic fixes and GPS positions are reported. Note that these discrepancies are all inside the “3σ” rule, i.e., less than three times the standard deviation of the position estimates (Table 1), however, they are visiting the boundaries relatively often. Fig. 5 reports the estimated trajectory with the EKF. Note that the EKF is initialized using also the GPS signal, then GPS is switched off from the EKF input. Fig. 6 reports the error between the EKF estimate and the GPS ground truth, together with the time instants in which an acoustic fix is received. One can observe that indeed the EKF navigation has a relevant drift when only INS data are used, and that the acoustic fixes are sufficient at beating the error down to acceptable levels. The acoustic fixes come with quite large intervals, due to the communication scheme behind their transmission: often packets are lost (between 30% and 70% packet loss was experienced), and the round robin scheme implies a delay that is proportional to the number of nodes. The largest error in the data (~40m) is registered in correspondence of the largest interval between two consecutive acoustic fixes (~3 minutes). Note that, if only localization is the issue, the modems can operate as pingers at a much faster rate. However, if communication is interleaved with the localization process, the transmission delays reported here are much closer to those that could be experienced in a team operation. Finally, conclusions are given.

5. CONCLUSIONS

The results in terms of localization accuracy of a concurrent communication and localization acoustic scheme for application in the operation of AUV teams for distributed search, survey and surveillance, have been presented. The data reported show how acoustic fixes obtained from the communication process are indeed successful in limiting the drift of on-board INS navigation system. However, the concurrence of communication and localization, at least in our current implementation, coupled with the fluctuations of the acoustic channel, causes large intervals between consecutive acoustic fixes.
Fig. 4: Typhoon path from GPS data (thin blue line), acoustic fixes from the on-board USBL (black upward triangles), corresponding GPS position (red downward triangles)

<table>
<thead>
<tr>
<th>Fix #</th>
<th>Error (m)</th>
<th>Fix #</th>
<th>Error (m)</th>
<th>Fix #</th>
<th>Error (m)</th>
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<tbody>
<tr>
<td>1</td>
<td>2.3</td>
<td>7</td>
<td>1.9</td>
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<td>3.4</td>
<td>8</td>
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<td>14</td>
<td>4.3</td>
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<td>8.2</td>
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<td>15</td>
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</tr>
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<td>10</td>
<td>8.5</td>
<td>16</td>
<td>11.6</td>
</tr>
<tr>
<td>5</td>
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<td>17</td>
<td>1.3</td>
</tr>
<tr>
<td>6</td>
<td>2.9</td>
<td>12</td>
<td>5.9</td>
<td>18</td>
<td>1.2</td>
</tr>
</tbody>
</table>

Table 2: discrepancy between the GPS measured positions and the acoustic fixes

Fig. 5: Left: GPS measured path (thick purple line), EKF estimated path (thin blue line), acoustic fixes (red diamonds); Right: discrepancy as a function of mission time between the GPS measurements and the EKF estimate, shown for a 20 minutes time window; the green diamonds correspond to the time instants in which an acoustic fix is received. The acoustic fixes succeed in beating down to zero the otherwise increasing drift in horizontal position estimate with INS only
ACKNOWLEDGEMENTS

This work has been partially supported by PAR FAS REGIONE TOSCANA Linea di Azione 1.1.a.3, project Thesaurus (http://thesaurus.isti.cnr.it), and by Italian Ministry of Research, Research project of National Interest MARIS. The Authors are indebted to John Potter, CMRE Scientist in Charge, CommsNet13 research cruise, to all the CMRE staff that supported them during CommsNet13, and to Officers and Sailors of the R/V Alliance.

REFERENCES

Session 12

Experimental and modelling validation of target strength measurements

Organizers: Duncan Williams, David Nunn and Alan Hunter
CONCEPTS FOR RELIABLE TARGET ECHO STRENGTH MEASUREMENTS AND IMPROVED TARGET REPRESENTATION

David Prowse

Abstract: Target Echo Strength (TES) is a fundamental component of the active sonar equation. Knowledge of TES signatures is needed in order to understand the performance and design drivers of active sonar detection systems, to develop and implement target detection and classification algorithms and exploit TES signatures to either maximise target detection, or remain stealthy. It is therefore important that TES is measured as reliably and that the origin of the reflected energy can be identified. Through simulation of TES measurement scenarios, this paper considers the advantages of a 2 dimensional array TES measurement system over a 1 dimensional single beam system in its ability to conduct reliable, accurate TES measurements of extended targets are demonstrated.

Keywords: Target Echo Strength, TES, Acoustic Measurement, Acoustic Modelling, Reverberation, Sonar.

1. INTRODUCTION

Target Echo Strength (TES) is a fundamental component of the active sonar equation. Knowledge of TES signatures is needed in order to understand the performance and design drivers of active sonar detection systems, to develop and implement target detection and classification algorithms and exploit TES signatures to either maximise target detection, or remain stealthy. It is therefore important that TES is measured as reliably and that the origin of the reflected energy can be identified.

Through simulation of TES measurement scenarios, this paper considers the advantages of a 2 dimensional array TES measurement system. The reliability, accuracy and scatterer localisation capability is compared with an equivalent 1 dimensional single beam system.
2. STUDY BACKGROUND

This study performs modelling of a mono-static TES measurement system consisting of a square 8x8 element transmit and receive array, with element spacing of 0.015m. The transmit frequency of the system is nominally 50kHz. The receive beam pattern is shown in Fig. 1. At this frequency, the -3dB beam width of the receive beam is 12.8deg in both azimuth and elevation for both transmit and receive.

![Fig. 1: Beampattern of modelled sonar array.](image)

The transmit and receive sonar is suspended at mid water depth in 100m of water. A target scatterer of interest is positioned at a range of 225m. The beamwidth is designed such that the 0º beam intersects the reverberation boundaries at a range greater than the target echo is received in order that boundary reverberation is minimised.

The 1D single beam system considers the TES of the scatterer is defined to be -20dB. The measurement configuration is illustrated in Fig. 2.

![Fig. 2: Modelled measurement setup.](image)

3. EFFECT OF ALIGNMENT ON TES MEASUREMENT OF POINT SCATTERER

3.1. Ideal Alignment
When the measurement system is perfectly aligned (i.e. the azimuth and elevation of the target are both 0º with respect to the centre of the sonar array), the centre beam processed signal is shown in Fig. 3. In the ideally aligned scenario, the 1D and 2D results are identical. The top plot shows the time series received signal. The signal has been adjusted for source level, receive sensitivity and transmission loss and is therefore presented as scattering strength (i.e. TES). The target echo is visible at a time delay of 300ms. Reverberation from the surface and bottom boundaries are also present, which cause an increase in the background levels. The target has been positioned so that it is in a region of comparatively low reverberation, i.e. at a range less than the intersection of the main beam and the reverberation boundaries. The bottom plot shows the time history over 200 observations. In the ideally aligned scenario, the 1D and 2D results are identical.

![Fig. 3: Received signal for an ideally aligned measurement. (Top) A-scan, (Bottom) History over 200 observations.](image)

The ideally aligned measurement scenario estimates the median TES of the target to be -19.98dB, with a standard deviation of 0.08dB. Fig. 4 shows the time history of TES measurements of the target over 200 observations and the distribution of observed TES values. The variance in the TES estimation is due to the reverberation interfering with the reflected target signal. The accuracy of TES estimation, even for an ideally aligned measurement, is therefore limited by the characteristics of the noise and reverberation background it is measured against. The limitation on accuracy will be dependent on signal to noise ratio of the target echo, with greater signal to noise ratios being less affected.
A single beam measurement system therefore can provide TES estimates to the accuracy enabled by the background characteristics for a point target scatterer. However, when considering a non-ideally aligned measurement, or an extended target object, the single beam approach becomes less accurate and limited in its ability to resolve scatterers.

### 3.2. Non ideal alignment

In practice, ideal alignment is difficult to achieve and should be considered as a special case. Movement of target and sonar orientation due to tidal currents, movement of arrays and/or target to achieve variation in observation angle and non-isospeed sound propagation will affect the alignment of the measurement system, and target. As the measurement configuration becomes non-aligned, the peak response of transmit and receive beams no longer point directly at the target. The non-alignment therefore reduces the effective Source Level and Receive Sensitivity in the direction of the target leading to a less accurate TES estimate if not corrected for.

The impact of non-ideal alignment on TES estimation on a 1D measurement system is shown in Fig. 5, where the elevation of the measurement arrays is varied from -4 deg to +4º, and the azimuth is simultaneously varied from -10º to +10º. This causes two main effects that degrade the accuracy of the TES estimation. These effects are the reduction in received reflected signal from the target due to not being at the peak beam response; and the increased levels of boundary reverberation when the sonar arrays are tilted towards the surface and bottom boundaries.
The visible impact of these effects for a 1D measurement system is that the target becomes increasingly difficult to distinguish against the background when the measurement is not aligned. At the maximum angles considered above (±4º elevation, ±10º azimuth), the target is no longer visible and reliable TES estimates are not possible.

A 2D system has some key advantages in this scenario. Firstly, the ability to form a number of receive beams in azimuth and elevation enable a direct and real time indication of the alignment of the measurement. This will enable corrective action to be performed to reduce the alignment errors in the measurement. Furthermore, because a direct measurement of relative angle is available, it is possible to correct the measurement for the beam response and regain an accurate TES estimates. Fig. 6 shows a typical 2D beamformer output from the configuration above for various alignments.

The distribution of TES estimates for the scenario presented in Fig. 5 are shown in Fig. 7. The TES estimates for the 1D systems are severely affected by non alignment, with TES being consistently underestimated. The median TES estimate from the 1D system is -22.02dB, with a standard deviation of 2.65dB. The 1D system tends to under estimate TES, with excursions as large at 10dB observed. The 2D system is successful in identifying the direction of the target and correcting for the beam response allowing for a reliable TES estimate to be made. The median TES estimate obtained from the 2D system is -20.04dB with a standard deviation of 0.46dB.
4. EXTENDED TARGET MEASUREMENTS

The analysis above considers the simple case of a point scatterer. In most practical cases, the target of interest has some spatial or temporal extent (either through the physical dimensions of the target or extended echoes caused by elastic scattering modes). To aid TES signature analysis and subsequent usage, it is important to identify the origin of the individual scatterers within an extended target.

Consider a target consisting of 5 scatterers distributed along a line target. For this type of measurement, the variation of TES with observation angle is often of interest. In the analysis presented below, the measurement sonar arrays are moved in a circle around the centre point of a 5 point scatter line target. Fig. 8 shows the measurement set up, with the red crosses indicating the position of the scatterers and the coloured circle and dashes representing the relative position of the measurement sonar.

The signal received by the 1D system for the best case alignment (i.e. sonar is aligned on the centre point of the target) is shown in Fig. 9 (left) for a 360° target rotation.
However, even in this aligned case, due to the extended nature of the target, any scatterers not centrally positioned are affected by the effects of non-alignment discussed above. TES of these scatterers will therefore be incorrectly estimated unless and appropriate measurement and processing is applied. This is apparent in Fig. 9 by the fading of the outermost target highlights and the reduction in return at 90° in the left hand plot. Conversely, the 2D system (Fig. 9, right) displays consistent TES returns for all scatterers at all aspects.

![Waterfall plots for (left) 1D system (right) 2D system. The 2D system displays consistent TES returns as target aspect angle approaches 90°.](image)

The PTS and ITS of the target as measured by a 1D system is shown in Fig. 10. The difference between 1D and 2D in PTS is limited due to the centre scatterer always being well aligned to the sonar. The ITS estimated by the 1D system is underestimated by up to 4dB at aspects near 90° and 270°, whereas the 2D system provides robust TES estimates across all angles.

![PTS and ITS of an extended target for 1D and 2D systems. The ITS estimated by the 1D system (red crosses) is underestimated by up to 4dB as target aspect angle approaches 90° and 270°.](image)
The 1D system has a further limitation. At 90° orientation, i.e. where the longest axis is parallel with the sonar, the scatterers along the target length are unable to be resolved and therefore there is no information on the origin of each scatterer on the target.

![Plot of position estimates from 2D system. – 0°, 25°, 50° & 90° target angles showing the ability of a 2D measurement system to resolve extended targets.](image_url)

Using a 2D system, the extended nature of the target can be observed in the beam plot. This provides direct confirmation that the target is contained within the field of view of the measurement system and ensures that the extremities of the target are not truncated. Furthermore, due to the multiple beams, it is possible to obtain a bearing estimation for each scatterer. This information is important and necessary when generating high fidelity highlight models for use in detailed modelling.

**5. SUMMARY**

A TES measurement system, consisting of a 2 dimensional sonar array and associated processing, has been modelled. The 2D measurement system has a range of advantages in its ability to accurately and reliably measure TES when compared to an equivalent 1D system. These advantages include direct measurement of the relative target angle and confirmation that the target is within the field of view of the measurement system; correction for off axis beam response enabling significantly improved reliability of TES estimates; and the ability to resolve and localise scatterers from extended targets.
MODEL TANK EXPERIMENTS AND USING A RANDOM NOISE FIELD TO DETERMINE SCATTERING PROPERTIES OF AN OBJECT

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Abstract: We present hydrophone measurements taken at the periphery of a model tank to demonstrate that the ambient noise field in the model tank can be transformed into coherent radiation suitable for probing the tank. This suggests that one might determine the structural properties of an object in an analogous way. In particular, we derive a method to estimate the structural or surface impedance matrix (or equivalently the inverse of the structural Green’s function) for an elastic body by placing it in an encompassing, spatially random noise field and cross-correlating pressure and normal-velocity measurements taken on its surface. A numerical experiment is then presented demonstrating that the scattered field obtained from this noise correlation based structural impedance agrees with standard scattering theory.

Keywords: Noise correlation, scattering, model tank experiment, structural impedance
1. INTRODUCTION

Determining the scattering properties of a complex object with a large number of internal degrees of freedom is a very costly endeavor (see [1] and papers referenced therein). Potential alternative approaches not requiring either exhaustive computations or measurements may be useful. Recently noise correlation based processing previously has applied to studying acoustic propagation in an assortment of scenarios (see for example [2-6] and the references contained therein) may provide such an alternative.

In this paper we examine the potential of utilizing noise fields for studying the structural and scattering properties of objects. The first example of an object is a 5 m diameter fish tank in the basement of a building that whose internal vibrations originate from the building ambient noise. Experimental results are presented that verify that the correlation of the ambient noise between hydrophones at the periphery of the tank results in signals that emulate active probing of the tank by deterministic signals. The second example is a theory/simulation result for an elastic shell placed in an ambient noise field. The results are in agreement with those obtain from scattering theory using conventional structural acoustics. Taken together, these examples point to the potential of utilizing ambient noise to probe structures.

2. NOISE MODEL TANK EXPERIMENT

A water tank of 5 m diameter and about 1 m deep was instrumented [7] with 16 hydrophones on its periphery to measure ambient noise within the tank with the only source of the noise being the ambient building vibrations (see schematic of Fig 1.). As shown there were also 8 sources on the periphery. The goal was to show that by just using ambient noise measurements and processing we could essentially reproduce the transfer functions obtained by the active source measurements.

Noise correlation theory [2-6] basically states that the time domain Green’s function between to positions is related to the cross correlation of noise between two sensors (1 and 2) through the relation,

\[-dC/dt \sim G(t)-G(-t),\]  

yielding the result that a passive measurement and subsequent processing of data at two sensors emulates the active transmission from a source at position 1 to sensor 2 and the reverse. The experiment then consisted of correlating the noise field between all receivers thereby obtaining the transfer functions between all receivers as if in each pair, one of the receivers was the source. This result is compared to an active implementation of the experiment in which there were only 8 sources to use to measure the transfer functions between the sources and the receivers. One complication of the experiment was that the noise field was strongest below 1500 Hz whereas we were using sources in the 10 kHz range. The noise measurement result was obtained by taking advantage of the rotational symmetry of the sensor configuration in that the correlations between rotated pairs were incoherently summed. The results in Fig. 2 unambiguously shows that, indeed, these
Fig. 1: Schematic of top view water tank that had a 5 m diameter and was 1 m deep. There were 16 receivers and 8 sources. The tank was in the basement of a building and the passive part of the experiment simply utilized the ambient building noise.

Green’s functions are embedded and recoverable from the noise field. Further, the correlation integration time to yield these results is consistent with the standard result that the correlation peak build-up is proportional to the square root of the time-bandwidth product.

To summarize this part, we have shown an example of probing the internal structure of an object using ambient noise. Theory states that the measured transfer functions would change if there were additional structures within the tank so that these type of measurements may be useful for either determining what these structures are, or more likely, for detecting change (often referred to as structural health monitoring).

3. SCATTERING PROPERTIES FROM NOISE CORRELATION PROCESSING

Extending the ideas of the work shown in Section 2, here we derive a method to estimate the structural or surface impedance matrix (or equivalently, the inverse of the structural Green’s function) for an elastic body by placing it in an encompassing and spatially random noise field and cross-correlating pressure and normal velocity measurements taken on its surface [2-6]. The structural impedance matrix represents all the internal elastic interactions that determine the resulting scattered field from incident field impinging on an object. A schematic of the methodology summarized from the literature [8-13] is shown in Fig. 3. The noise correlation method applied to this problem directly gives
the structural acoustic impedance matrix that is representative of the response of an object to an applied excitation as if the object was placed in a vacuum. One sees that with this formulation as illustrated in Fig. 3, given the structural impedance matrix $Z_s$ that is totally representative of the internal elastic structure, one can determine the scattered field for an object placed in any external medium (actually not restricted to a fluid medium, see [11]). Reference [8] derives an expression for the surface impedance matrix (in the frequency domain) from noise measurements,

$$Z_s = -\langle pp^H \rangle \langle vp^H \rangle^{-1}$$ (2)
Fig. 3: Three impedances are required to describe the scattering problem using either the equivalent source method (ESM) or the Helmholtz integral equation: $Z_a$, $Z_i$, and $Z_s$, which are the acoustic, the internal, and the structural impedances, respectively. The acoustic and the internal impedances characterize the vibration of the outer fluid outside and within the body surface and thus only depend on the body shape and the outer fluid property. The structural impedance characterizes the response of the elastic body in vacuum, thus only depends on the object’s structural parameters.

measurement is for the object placed in the specific medium of the experiments whereas the correlation of noise quantities result in the structural impedance matrix that is independent of the outer medium. The general result for an arbitrary outer medium then follows the methodology indicated in Fig 3.

We have proceeded through this methodology for scattering from a cylindrical shell using a COMSOL implementation of Eq. 2 combined with ESM and from the standard analytic scattering theory/Helmholtz Integral method. Equation 2 is implemented in COMSOL using a set of random, isotropic plane wave. Figure 4 shows the COMSOL results for the structural impedance that will be combined with ESM to give the scattered field results since it is the same as the COMSOL result. Also, we have performed calculations encompassing the resonances of the shell that verify that the noise correlation method accurately includes these regions. Finally, Fig 5 shows the scattering results for the COMSOL numerical experiment using noise correlations overlying identical results from the standard analytical solution to the problem. We used this simple object because the analytic result is known. However, the applicability of the noise correlation procedure is independent of the complexity of the object.
COMSOL ESTIMATION OF $Z_s$
with Random Plane Waves

Fig. 4: The structural impedance matrix for a cylindrical shell. The axis of the matrix contour plot corresponding to Eq. 2 is the angles that correspond to specific nodal points on the surface of the cylinder used in the COMSOL numerical realizations. This result is then combined with the ESM to give the scattered field as in Fig. 5.
4. CONCLUSIONS

We have shown examples of using random noise to ascertain deterministic properties of structures. By measuring the building generated ambient noise in a large model tank, we were able to measure the deterministic acoustic propagation between passive sensors thereby replacing active probing with passive monitoring. Further, we have shown with a numerical experiment that the structural impedance of an object can be determined by measuring the pressure and normal velocity at a set of points on the surface of the object that is placed in an external noise field. This particular processing isolates the structural properties of the object independent of the external medium. Taken together, these results suggest that the utilization of ambient noise through measurement and correlation based processing may provide as assortment of opportunities to develop measurement/modeling procedures beyond those that are presently limited by our inability to model the propagation characteristics of complicated media.

5. ACKNOWLEDGEMENTS

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ON THE DESIGN AND CONSTRUCTION OF DRIFTING-MINE TEST TARGETS FOR SONAR, RADAR AND ELECTRO-OPTICAL DETECTION EXPERIMENTS

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Abstract: The timely detection of small hazardous objects at the sea surface, such as drifting mines, is challenging for ship-mounted sensor systems, both for underwater sensor systems like sonar and above-water sensor systems like radar and electro-optics (lidar, infrared/visual cameras). This is due to the low target echo strength and radar/lidar cross-section of partly submerged objects at small grazing angles, which are also intermittently shielded by waves and their response is being hidden in significant surface reverberation. In 2009-2010, the feasibility of ship-based detection by state-of-the-art sensor technology was successfully assessed using specially designed drifting-mine test targets. In this paper, we look at the test target requirements for different observation technologies, and focus on the design and construction of the target objects. For illustration of the suitability of the test targets and available sensor technology, some sea-trial results are included for visual and IR detection above water, and for sonar detection under water.

Keywords: Drifting mines, test target design, target detection experiments.

1. INTRODUCTION

Small floating/drifting objects pose a severe threat to surface ships, especially if the objects are explosives. Timely detection is crucial for conducting adequate evasive manoeuvres. There are roughly two strategies for the detection of drifting objects: above-water sensors and underwater sensors. In order to make ships self-sufficient, there is a preference to connect all sensors to the ship, resulting in low grazing angles for the detection of objects at the sea surface and well ahead of the ship. Consequently, the
detection of drifting mines is very difficult for both above-water and underwater sensors due to surface effects caused by the sea-state (wave occlusion and scattering).

What if we combine the use of above-water and underwater sensors, in which above-water sensors compensate for low detection probability by their underwater counterparts, or vice-versa? To put this idea to the test, we studied state-of-the-art high-frequent sonar systems for underwater detection and state-of-the-art radar, lidar (laser) and EO systems for above-water detection. A theoretical feasibility study was carried out to assess whether radar technology is suitable for the detection of drifting mines [1]. Experiments were performed using specially constructed test objects, including coastal measurements for lidar [2] and EO (IR and visual) [1] and sonar tests at sea [2][this paper].

In the present paper, special attention will be given to the construction of the drifting test targets for underwater detection (sonar) and above-water detection (EO).

2. DRIFTING-MINE THREAT

A shortfall of conventional naval mine-countermeasure (MCM) capability is the unsuitability to deal effectively with objects drifting (floating) freely at the sea surface. Although outlawed by the 1907 Convention of The Hague, it should not be ruled out that purpose-built drifting mines are deployed intentionally in times of war. However, more likely, the drifting objects will be sea mines that have become detached from their moorings due to fatigue, corrosion or (mechanical sweep) cutting by MCM vessels (Fig. 1). Although their firing mechanism should become inactive when detached (using a pressure or cable-strain sensor), they may still be a threat due to collisions with ships. Furthermore, improvised explosive devices (IED), e.g. drifting barrels put in the water by terrorists, are also becoming a serious threat.

Fig. 1: a) A drifting mine in the Persian Gulf (Proc. USNI, February 2011). b) Disposal of a WW2 drifting mine (NL Defence magazine “Alle Hens”, May 2002).

Old-fashioned sea mines, as shown in Fig. 1, typically have a metal shell and are equipped with horns for activation by physical contact. More modern sea mines may have a smooth plastic shell and use internal pressure, acoustic and/or electro-magnetic sensors for activation.
3. CONSIDERED OBSERVATION TECHNOLOGIES

Since drifting mines may be detectable both above and under water, with the preferred observation technology depending on the submerged fraction, the mine’s composition and the environmental conditions, it makes sense to consider both above-water and underwater observation technologies. The detection geometry is sketched in Fig. 2.

![Detection Geometry](image)

**Fig. 2: The detection geometry for detection of a drifting mine by ship-mounted and remote sensors.**

In general, drifting mines are largely submerged (70-80%), rendering active ship-mounted detection more probable underwater, with active sonar, than above water, with radar or lidar. Active technologies become more attractive when the grazing angle increases, e.g. by detection from a (ship-based) remotely-operated/unmanned/autonomous underwater/aerial vehicle (ROV, AUV, UAV). For mast-mounted above-water observation technologies, passive detection with infrared (IR) camera technology is to be preferred [1].

4. SUITABLE DRIFTING MINE-LIKE OBJECTS

Drifting mines are most probably released moored mines, which typically have a (nearly) spherical shape. Furthermore, active above-water radar detection was analysed to be most difficult for smooth spherical (and largely submerged) objects [1]. Therefore, it was decided to use smooth spherical objects as drifting mine-like objects. Suitable objects were obtained from the Belgian Navy, see Fig. 3. These discarded moored ED34 exercise mines were used without modifications for passive IR detection tests [1] (see also §5.1), but were adapted for active lidar/radar and sonar tests. The air-filled 83-kg iron spheres have an outer diameter of 0.815 m.
Fig. 3: The ED34 exercise mine spheres, in their original colour scheme and configuration.

For the active above-water detection tests, one of the spheres was stripped of all of its protrusions except for a single eyebolt needed for loading and deployment. The polished metal sphere was consequently painted black, see Fig. 4.

Fig. 4: The exercise mine sphere adapted for active above-water detection testing (smooth on top, weight below).

For the active underwater detection tests, the other sphere was also stripped of all of its protrusions except for a single eyebolt, but now only needed for deployment, not loading. No weight could be attached under the sphere as that would influence the detection probability significantly. The only suitable way to load the sphere was by adding weight inside the sphere. Hereto, the sphere was opened on its top-side, by creating a water-tight round hatch next to the handling eyebolt. Furthermore, the sphere was polished and coated in a bright signal-orange colour to aid relocation. The relocation was also more important than for the black sphere, because the black sphere was (loosely) attached to a rope under water, while the orange sphere was truly free-drifting in the sea. Therefore, the orange sphere was also equipped with a radar reflector and big pink balloons (Fig. 5). It should be noted that despite the radar reflector, the sphere was not visible on the ship’s surface radar, indicating the difficulties for mast-mounted radar detection.
The internal loading of the orange sphere was a complicated exercise. The small hatch in the sphere did not allow to insert a big weight as used to load the black sphere (Fig. 4, left), and we also did not want the weight to make contact with the sphere as this would influence the acoustic and structural properties of the sphere undesirably. Eventually, it was decided to pour in melted lead, isolated by a thick layer of poly-urethane (PUR) foam. However, this was easier said than done.

As is shown in Fig. 6, a steel pipe, with protrusions at its lower end, was mounted inside the sphere (fixed at the hatch opening). Consequently, ~155 kg of liquid lead was poured in and cooled down to its solid state (this should actually have been done before painting, as the heat burned off the paint at the bottom side). Next, the solid lead was lifted ~10 cm by lifting the steel pipe. Finally, a 2-component PUR-25 mixture (25 litre/kg after full expansion) was poured through the pipe, filling the sphere from below the lead weight. This procedure created a low centre of gravity, providing stability at sea.

After hardening of the PUR foam, the upper part of the steel pipe was removed and the sphere was balanced in the TNO basin using a 10-kg lead bar, fixated by some additional PUR foam from a spray can, see Fig. 7. The total added weight of ~178 kg (incl. foam, steel pipe, hatch construction, and mast with GPS and radar reflector) submerged the sphere by about 80% of its diameter in sea water.
Fig. 7: Balancing of the orange sphere in TNO’s freshwater basin using a lead bar fixated by additional PUR foam from a spray can.

5. SOME TRIAL RESULTS

5.1. Electro-optical detection

Fig. 8 shows some passive EO detection results at ~900 m distance of the (unmodified) drifting mine sphere for a visual (black & white) camera and a mid-wave infrared (MWIR) camera. The sphere was loaded by a 127-kg external weight, submerging it for about 65% of its diameter in sea water. More details of this 2009 experiment are reported in [1]. It is clear that the contrast is much better for the (MW)IR camera than for the visual camera. This is due to the fact that at low grazing angles, the flat sea surface is ‘coloured’ by the (reflected) sky’s temperature instead of its own water temperature, whereas the IR camera sees the mine sphere and other protrusions (buoy, boat, …) by their own radiated temperature and reflected sea temperature too. So even when the mine sphere is in thermal equilibrium with the sea surface, there will still be contrast. The contrast decreases for cloudy skies and high sea states. However, the detection can be further improved by video processing, i.e. exploiting the different dynamics of a drifting mine and wave tops.

Fig. 8: Passive detection of a drifting mine sphere (see Fig. 3), on the left using a visual B&W camera, on the right using an MWIR camera. The distance between camera and sphere is the same for both cases (~900 m). The wind speed was 7-8 m/s.
5.2. Sonar detection

Fig. 9 shows the setup of the sonar target strength (TS) measurements of the orange sphere. Combined measurements of a reference target (glass sphere) with a known TS using the same high-frequent (HF) ship sonar enabled the statistical determination of the absolute TS of the mine sphere for the applied HF sonar mode: $-21 \pm 6$ dB, compared to a theoretical TS of $-14$ dB for an air-filled sphere of 80 cm internal diameter. The variation is due to the variable degree of submersion at the wavy surface. At maximum degree of submersion (~100%), the measured target strength is then about $-15$ dB, which comes close to the theoretical value. However, the objective was actually not to mimic an air-filled sphere, but to have a drifting sonar target with a constant, known and omni-directional TS. Quite likely, a real drifting mine will have a smaller TS due to its contents.

![Fig. 9: Setup for measurement of the sonar target strength (TS) of the drifting mine sphere. The sound-speed profile was approx. uniform. The sea state was approx. 3-4.](image)

After the static TS measurements, dynamic detection experiments were performed with both the ship’s hull-mounted sonar and its remotely-operated self-propelled variable-depth sonar, see Fig. 10 for the run geometry. The ROV is to be replaced by a stand-off operating AUV in the future. More details of this 2010 experiment are reported in [2].

![Fig. 10: Run geometry for the drifting-mine detection experiment, using both the ship’s hull-mounted sonar and its variable depth sonar, both moving at a speed of a few knots.](image)

A sonar image of the ship’s hull-mounted sonar is shown in Fig. 11. The figure shows an automatic detection of the drifting mine sphere at several ship’s lengths distance. This example is for a strong detection as the zoom image on the right shows clear sidelobes due to saturation. However, due to the target regularly disappearing behind the waves (wave occlusion), the detections are intermittent and difficult to track automatically. The sonar processing would have to be optimized for this intermittent behaviour. A possible approach is scan-to-scan integration and de-correlation, thus collecting reliable statistics by increasing the time-on-target (increasing reaction time). Another idea is to train a classifier on the dynamical behaviour of drifting mines.
Fig. 11: A sonar image of the ship’s hull-mounted sonar (HF mode). The right image zooms in on an automatic detection of the drifting mine target on the left.

6. CONCLUSION

Suitable drifting-mine test targets were identified, adapted and used for detection tests using above-water sensors (EO, lidar, radar) and underwater sensors (sonar). Sensor fusion for above- and underwater sensors is relevant for detection of drifting objects as they are located at the air-sea interface. Mid-wave infrared (MWIR) cameras seem suitable for mast-mounted detection and thus for self-protection of high-value units. However, underwater detection requires high-frequent sonar, either hull-mounted or stand-off AUV-mounted, which implies the necessity of specialized MCM assets.

Important factors for the detection of drifting mines are the degree of submersion and the grazing angle of the sensor beam with the sea surface. Furthermore, because the acoustic impedance jump from water to the mine’s interior is important for sonar detection, a substantial effort was made in this study to create a test target with a constant, known and omni-directional target strength. A practical explanation for creating such a target is included in this paper.

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IMPROVED MODELING ACCURACY OF THE ELASTIC OBJECT RESPONSE BY INCLUSION OF HIGHER ORDER SCATTERING

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Abstract: Current methods of automatic target detection and classification using low- to mid-frequency sonar are based on correlating the measured Target In the Environment Response (TIER) of potential targets with the TIERs in a database of known objects. Obtaining experimental TIER measurements of proud, partially and fully buried objects to populate the database is an expensive undertaking. Using numerical predictions of the TIER instead of experimentally gathered data is a potential efficient alternative. In addition, a numerical model also offers insights into the physics involved as an added advantage. Previously, an efficient numerical hybrid method based on Finite Element and Helmholtz Kirchhoff Integral (HKI) models was developed for prediction of the TIER of arbitrarily positioned axially symmetric objects. This hybrid model was successfully tested and validated for a number of cases. However, the main limitations of this hybrid method (besides being restricted to axially symmetric targets) is the neglect of higher order re-scattering between the target and the sediment, and the inability to account for discontinuities or changes in the sound speed and density of the medium surrounding the target surface (unless such changes are perpendicular to the axis of symmetry). For certain target/environment combinations, comparison of measurement and model results suggest that higher order re-scattering contributes significantly to some features of the acoustic response. The hybrid model was extended to include higher order re-scattering by successive iterations of running the original model. As part of the approach a novel non-singular HKI method was implemented and successfully extended to work with non-homogenous media. The resulting hybrid model that includes higher order re-scattering was validated against the results of an experimental test-setup for which higher order re-scattering effects are shown to be relevant.

Keywords: Target In Environment Response, Multiple Scattering, Elastic Target Response
1. INTRODUCTION

The development and training of automatic target detection and classification techniques for mines and unexploded ordnance (UXO) shells based on synthetic aperture sonar (SAS) imaging in general requires the so-called Target In the Environment Response (TIER) information of these objects to be known. The results presented here are part of a study aimed at the development, testing and validation of efficient computational techniques for the modeling of the target in the environment response (TIER) of proud, partially and fully buried axially symmetric mine-like objects and UXOs. The restriction to axisymmetric target shapes and the use of adequate approximations for treating the presence of the water-sediment interface make it possible to obtain TIER results for relevant situations without the need to resort to large parallel computing facilities [1,2,3].

The main limitations of the developed method (besides being restricted to axially symmetric targets) is the neglect of higher order re-scattering between the target and the sediment, and the inability to account for discontinuities or changes in the sound speed and density of the medium surrounding the target surface (unless such changes are perpendicular to the axis of symmetry). Validation with experimental data (from PondEx08, PondEx09 and PondEx10 [4,5,6]) and full 3-D FE models (developed by D.S. Burnett at NSWC PCD [3]) suggests that in many cases this effect does not play a major role in predicting the relevant target classification features of the acoustic templates. However, for certain target/environment combinations, comparison of measurement and model results suggest that higher order re-scattering contributes significantly to some features of the acoustic template. The results, for an experimental setup where higher order re-scattering effects are shown to be relevant are described below in section 4.

The hybrid model was extended to include higher order re-scattering by successive iterations of running the original model. As part of the approach a novel non-singular HKI method (see [6]) was implemented and successfully extended to work with non-homogenous media. The resulting hybrid model that includes higher order re-scattering was validated against the results of the experimental test-setup in section 4 for which higher order re-scattering effects are shown to be relevant.

2. HYBRID MODELING APPROACH

The modeling technique developed here is commonly referred to as a “hybrid model” and consists of three coupled models: An efficient model (usually based on analytical approaches) for expressing the field incident onto the target, a high fidelity finite element (FE) based model for the solution of the local target scattering component of the problem [1], and an efficient (approximate) model based on the Helmholtz-Kirchhoff boundary integral accounting for the propagation of the local target scattered field (i.e. the echo) to receivers located anywhere in the water column [1,2]. The coupling between the different models is schematically shown in Figure 1. The precise geometry of the experimental setups that are used for verification - including the relative position and orientation of target, source(s), and receiver(s) - is taken into account in the model [3].

The FE model is a linear frequency-domain structural acoustic model, applicable to axially symmetric target geometries subject to arbitrary (non-symmetric) incident fields. The three-dimensional (3-D) elastic displacement equations for the solid domains and the Helmholtz equation for the acoustic pressure in the fluid domains, are decomposed into a
set of independent 2-D FE equations using a Fourier expansion around the azimuthal coordinate of the cylindrical coordinate system in which the target geometry is described. The solutions to the independent 2D equations are referred to as azimuthal or circumferential modes. The number of circumferential modes that are used to represent the solution to the 3-D problem controls the accuracy of the decomposition. Each 2-D problem is substantially smaller than the original 3-D problem in terms of size of the system of equations and associated memory and computing requirements. Furthermore since the individual 2-D problems are independent of each other, their computation can be trivially parallelized.

The main limitations of the model presented in reference [1,2,3,4,6] (besides being restricted to axially symmetric targets) is the neglection of higher order re-scattering (higher order interaction) between the target and the sediment, and the inability to account for discontinuities or changes in the sound speed and density of the medium surrounding the target surface (unless such changes are perpendicular to the axis of symmetry).

3. HIGHER ORDER SCATTERING

In the present work, the hybrid model is extended to include higher order re-scattering by successive iterations of running the model. This process is illustrated in Figure 2. Note that while the primary interest of this research is in the TIER for objects interacting with the seafloor, the schematic shown in Figure 2 concerns the interaction of an object with the water-air interface. As part of the adopted approach to account for higher order re-scattering, a novel non-singular Helmholtz Kirchhoff Integral (HKI) method [7] is used to calculate the field incident on the target due to reflection of the (first order) scattered wave in the sediment water interface (modeling step (d) in Figure 2).

The non-singular HKI method described in reference [7] was developed for free field conditions. The method was extended to work for the case of a medium consisting of two coupled half-spaces by replacing the free field Green’s kernels and plane wave solutions
used in the HKI method by their associated solutions for a stratified layered medium. The non-singular HKI method for non-homogenous media that is thus obtained was successfully validated against FE results for 2D test cases and by using 3D test cases that have an analytical solution as a reference solution.

The non-singular HKI method is incorporated in the workflow of the hybrid model as followed: The incident field is evaluated on the targets surface (Figure 2(a)), and the result is decomposed into contributions to the individual circumferential modes. The response of the target to the incident field is calculated using the FE model (Figure 2(b)) for each circumferential mode. The combined response for all modes is used as input for the non-singular HKI method to obtain the scattered field after interaction with interface of the layered medium on the target surface (Figure 2(d)). The result of this ‘self-illumination’ is decomposed into circumferential modes which allows subsequent calculation of the response of this higher order scattered field with a second run of the FE model (Figure 2(e)). The response of the first order scattered field (Figure 2(b)) and second order scattered field (Figure 2(e)) are input for the far field HKI model yielding the TIER for first and second order of scattering in the far field (Figure 2(c) and Figure 2(f), respectively).

4. RESULTS

The classification features in the target echo are dependent not only on the target itself, but are influenced and modified by the environment surrounding the target. For this reason, it is of critical importance that the model be capable of simulating the echo of the target in all conditions, ranging from free field to proud, partially and fully buried and for arbitrary orientation of the target with respect to the fluid/sediment interface.
The TIER results for free field and proud targets were discussed and validated in [4,5]. For fully and partially/obliquely buried targets the TIER results were discussed and validated in [6]. For all these cases the model that was used only included the first order scattering. The model/data comparison suggests that in many cases higher order scattering does not play a major role in predicting the relevant target classification features of the acoustic templates. However for the case of an obliquely buried cylinder notable differences can be observed [6]. These differences might be caused by omission of higher order scattering. Furthermore, for certain target/environment geometries, comparison of measurement and model results suggest that higher order re-scattering contributes significantly to some features of the acoustic template. An example of such a setup involving an aluminum cylinder of length 0.0508 m and radius 0.0127 m that is positioned slanted relative to the water surface is schematically depicted in Figure 3. Experiments for this setup were carried out under controlled conditions in a tank at WSU [8].

Acoustic template plots of TS as a function of aspect angle and frequency for the aluminum cylinder suspended close to the water-air interface at an oblique angle of 34 degrees are shown in Figure 4. Illumination of the target from end-fire (90 degrees) as depicted on the right in Figure 3 causes sound to be reflected from the flat end cap towards the water-air surface in a direction nearly normal to the interface. The near normal incidence causes the sound to be reflected back to the target such that a second reflection on the end cap directs the sound back to the source/receiver combination causing a strong response for sound incident from one of the two end-fire directions. The strong response can be clearly seen in Figure 4(d) depicting the color template for the measured data for this setup. Figure 4(b) shows the color template for the model based on first order scattering only (obtained from model step Figure 2(c)). The strong response at end-fire (90 degrees) is fully absent in this plot.

![Fig 3: Example of target/environment geometry that can result in significant higher order re-scattering between the target and water/air interface (top).](image)

The new workflow including higher order re-scattering is used to predict the contribution of second order scattering. Figure 4(a) shows the contribution to acoustic template of the second order interaction with the cylinder (obtained by model step Figure 2(f)). Figure 4(c) shows the acoustic template based on the combination of first and second order scattering. By comparing the model results including the second order interaction (Figure 4(c)) and excluding the second order interaction (Figure 4(b)) with the measurement results (Figure 4(d)) it is clear that certain features of the acoustic template are due to (or heavily influenced by) the second order interaction with the object (i.e., re-scattering of the scattered field). Besides the response at 90 degrees end-fire for which a physical explanation is given above, a second strong response due to second order interaction with the cylinder is observed for higher frequencies at ca. 45 degrees. This response is thought to be related to excitation of surface waves on the cylinder surface.
which (depending on aspect angle and frequency) can efficiently radiate sound towards the water-air interface at normal incidence. After reflection in the water-air interface the path of the incident wave is traced back to the source. This interaction mechanism may also play a role for (half) buried objects that are positioned at an oblique angle with respect to the sediment surface. It remains to be verified whether the differences between model and measurement data observed in [6] can be (partially) explained by the effects of second order object interaction.

![Acoustic template plots](image)

**Fig 4**: Acoustic template plots of target strength (TS) as a function of aspect angle and frequency for the aluminum cylinder shown in Figure (3). The plots show the response due to 2nd order scattering (a) and 1st order scattering (b) and their sum (c). Comparison with the experimental data measured in the WSU tank (d) illustrates the importance of the contribution of the 2nd order, in particular at angles close to 90 and -45 degrees.

5. **CONCLUSIONS AND OUTLOOK**

The results presented above show that a significant step forward in the modeling and verification have been achieved. The inclusion of the effects of higher order re-scattering in the model offers new insights into the physical phenomena that play a role in target environment interaction. It is shown above that for certain cases, both model and experimental data suggest that these higher order scattering terms can have a significant influence on the features of the acoustic template.

The contribution of higher order re-scattering to the TS in case of half buried objects can be tested by applying the newly developed methods that are described here. Other cases where modeling the effects of higher order re-scattering are expected to be important involve, for instance, multiple objects or the presence of seabed roughness or ripples.

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Abstract: When attempting to detect and classify objects in an ocean environment, it has been shown that the surrounding environment, specifically the target’s deployment within that environment, greatly affects the measured acoustic response. While sea trials go a long ways towards extending the development of better classification systems, they are costly and ultimately only investigate the target at a handful of burial depths and ranges. There exists a need for robust models that can predict the target response with the required level of fidelity to result in positive classification outcomes. To accomplish this, however, reliable measurements and efficient models go hand-in-hand, one validating the other or vice versa. A specific example of this is presented here by examining the acoustic scattering from a solid cylinder in an ocean environment, partially buried with its axis at an oblique angle relative to the sand sediment. Plots of the measured target strength versus frequency and aspect angle reveal a number of interesting acoustic phenomena, some of which are explained using physical acoustics models. To help validate these measurements and offer further insight into the physical mechanisms involved, results from a numerical hybrid model are presented. This model, based on Finite Element and Helmholtz Kirchhoff Integral methods, has previously shown success in predicting the response of proud and partially buried targets with axes parallel to the sand sediment. To help validate the use of the hybrid model in this non-symmetric environment, a controlled tank experiment was conducted using a scaled version of the solid cylinder, mounted next to an air-water interface. The comparison between the hybrid model and tank measurements exposes potential areas where higher order re-scattering effects may become important, and is the topic of the following talk by M. J. J. Nijhof.

Keywords: Finite element modeling, physical acoustics models, acoustic scattering
1. INTRODUCTION

Detection and classification of objects in an ocean environment is a complex problem. Not only must you know how sound interacts with the elastic object, but also how it interacts and reverberates within the environment. Complicating matters is the exact orientation and position of the object within the environment, e.g. proud, partial or full burial. Previous work has shown that when the axis of the cylinder is tilted with respect to the sediment interface, there is a dramatic change in the broadside response [1]. Figure 1 illustrates this, comparing the data acquired in a test pond facility during PondEx09 for the scattering from a solid aluminum cylinder (length = 61.0 cm, diameter = 30.5 cm, length-to-diameter ratio equal to 2) in two different configurations within the sediment. The top panel depicts the absolute target strength (TS) as a function of frequency and aspect angle for the cylinder in a proud configuration (previously discussed in Ref. 2), while the bottom panel shows the results for the same cylinder partially buried with its axis tilted with respect to the sediment by 20 deg. (topic of Ref. 1). A hybrid FE/propagation model was developed that succeeded in capturing the dominant phenomena observed in the response near broadside [1]. Questions that still remain are: (1) How robust are these phenomena, i.e. are they still visible in a real ocean environment, (2) Can models be used to understand the physics driving these dominant phenomena, and (3) How well does the hybrid FE/propagation model predict the target response for aspect angles greater than broadside? These questions are addressed in what follows, while simultaneously illustrating the importance of experiment data and model validation.

Fig. 1: Target strength in dB as a function of aspect angle and frequency for a 61.0 cm long solid aluminium cylinder acquired during PondEx09. TOP: Cylinder proud on the flat sediment interface, 10 m range from source/receiver array. BOTTOM: Cylinder obliquely, partially buried, its axis making a 20 deg. angle with the flat sediment interface and 10 m range from the source/receiver.
2. EXPERIMENT DATA, MODEL PREDICTIONS AND PHYSICAL INTERPRETATION

2.1 Ocean Experiment

A solid aluminum cylinder was deployed during the Target and Reverberation Experiment (TREX13), conducted in the Gulf of Mexico off the coast of Panama City, FL in Spring 2013. The cylinder, measuring 90.4 cm in length and 30.5 cm in diameter (length-to-diameter ratio equal to 3), was partially buried (38% burial) with its axis having an 18 deg. tilt with respect to the sediment plane. It was located broadside and 15 m away from a linear rail system. The source and receiver array are mounted on a tower at a height of 3.8 m above the sediment. The tower transverses the 40 m long rail emitting a broadband chirp (1-30 kHz, center frequency 16 kHz) every 2.5 cm. This same rail/tower system has been used in previous experiments conducted in a test pond facility and in the Gulf of Mexico [2,3]. The synthetic aperture sonar (SAS) data is converted to absolute target strength (TS) and plotted as a function of frequency and aspect angle with respect to the incoming sound beam. This type of plot is called the acoustic template or acoustic color. Figure 2(a) shows the acoustic template for the partially, obliquely buried cylinder. For frequencies greater than 15 kHz, the broadside response has a distinct split at an aspect angle approximately equal to -5 deg. This same effect was observed in the controlled pond experiment shown in Figure 1(b). This splitting effect is a feature that appears to be robust enough to be observable in a reverberant ocean environment. In what follows we use models to understand the experimental data, and to dissect the physics driving the cylinder’s response.

2.2 Hybrid Finite Element – Propagation Model

A hybrid FE/propagation model was used to validate the experimental data in Fig. 2(a). The model decomposes the incident field numerically via a Fast Fourier Transform (FFT). This is applied directly to the surface Gauss points via the commercial FE software COMSOL, which is used to solve for the scattered pressure and pressure derivatives on a discrete set of points closely surrounding the cylinder [1]. These pressures and derivatives, combined with the appropriate Green’s function for a two-layered medium, are used in the discrete form of the Helmholtz integral to propagate the scattered field to the desired observation point in the far field. Figure 2(b) shows the TS predicted by the hybrid FE-propagation model for the same source/receiver geometry and cylinder orientation as that of Fig. 2(a). The model predicts the same splitting of the broadside response as was observed in the measured data.

2.3 Physical Acoustics Model

A physical acoustics model was developed in order to dissect the physics driving the cylinder’s response. Briefly, the model uses the expression for the far field scattered pressure from a finite length cylinder widely used in literature [4, 5]

\[
p_{\text{scat}} = \frac{af_{\text{scat}}}{2} \frac{e^{ikr}}{r},
\]

where the quantity \( f_{\text{scat}} \) is the backscattering form function for a cylinder of radius \( a \) and \( p_i \) is the amplitude of the incident pressure field. When the wavenumber-radius product \( ka \) is
large, the form function $f_{cyl}$ can be expressed as a sum over the available ray paths and a limited number of leaky wave elastic contributions,

$$f_{cyl} \approx e^{-i \pi /4} \frac{2 \sqrt[4]{kaL^2 \pi^2}}{a \pi} \left( f_{s,p} + f_{l,p} \right).$$ (2)

The subscript $S$ refers to the specular contribution, $l$ refers to the $l^{th}$ leaky wave elastic contribution, and $L$ is the cylinder length. The ray paths are denoted by the index $p$. The 4 ray paths that exist in this geometry are (1) a direct path to/from the cylinder, (2) a direct path to the cylinder, and a single bounce off the sediment on the return path (3) the reverse of path (2), and finally (4) a bounce off the interface before and after interacting with the cylinder. The derivation of the form functions $f_{s,p}$ and $f_{l,p}$ are beyond the scope of this paper, but the basis lies in the evaluation of the compact integral expressions given in Section 2 of Ref. 6, which are used to derive the form function for a rigid cylinder of finite length. The end result is a model that has included the effects of the finite length of the cylinder, the tilt with respect to the sediment interface, elastic mechanisms (specifically circumnavigating Rayleigh waves), the reflection coefficient of the sediment interface and an angular spreading factor. The results for the obliquely buried cylinder using this physical acoustics model are given in Figure 2(c). This model includes the important physics that are responsible for producing the splitting of the broadside response.

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Fig. 2: Target strength in dB as a function of aspect angle and frequency for a solid aluminium cylinder, broadside to the source/receiver, obliquely buried (38%) with its axis making an 18 deg. angle with the flat sediment interface. TOP: Data acquired during TREX13. MIDDLE: Hybrid FE/propagation model results. BOTTOM: Physical acoustics model results.
3. VALIDATION OF MODELS USING TANK EXPERIMENT DATA

Thus far we have demonstrated the effectiveness of the models at predicting the cylinder's response near broadside, and used them to verify the data measured in the ocean environment. In order to investigate the use of the hybrid FE-propagation model as a viable tool for predicting the target response for aspect angles much greater than broadside, a tank experiment was designed and conducted at Washington State University. A scaled version of the solid aluminum cylinder used in the PondEx09 experiments (Fig. 1) was manufactured, measuring 50.80 mm in length and 25.40 mm in diameter (length-to-diameter ratio equal to 2). The cylinder was mounted as close as possible to the air-water interface without actually breaking the surface. The cylinder was tilted so that its axis made an angle of 34 deg. with respect to the flat interface. The incoming sound beam has a grazing angle of 19.5 deg., also relative to the flat interface. The sound backscattered to the transducer (at a slant range equal to 2.52 m) was measured as the cylinder was rotated about the vertical azimuth. The angle convention is such that 0 deg. corresponds to broadside incidence, -90 deg. has the cylinder end farthest from the interface pointed toward the source/receiver, and +90 deg. the opposite. Figure 3(a) shows the TS for this experiment, which has been scaled to absolute values using a technique outlined in Ref. [7]. Figure 3(b) shows the TS predicted by the hybrid FE/propagation model. In addition to predicting the same splitting effect at broadside, the model also accurately predicts the measured response for aspect angles between -50 and -70 deg. and between 40 and 60 deg. There is a large discrepancy however, at 90 deg. and -40 deg., where the model greatly under-predicts the target response. At these aspect angles, the target is oriented in such a way that there exists a ray path for multiple scattering between the cylinder and the air-water interface. This is an effect that cannot be captured by the current hybrid FE/propagation model.

Fig. 3: Target strength in dB as a function of aspect angle and frequency for a solid aluminium cylinder, next to an air/water interface and tilted so that its axis makes a 34 deg. angle with the interface. LEFT: Data acquired in the WSU tank facility. RIGHT: Hybrid FE/propagation model results.
4. SUMMARY

The acoustic response of a cylinder partially buried at an oblique angle was investigated. Measured data acquired in a test pond and a reverberant ocean environment shows a split of the broadside response. A hybrid FE/propagation model and physical acoustics model were used to understand the experimental results near broadside and help to interpret the physics driving the target response. A scaled tank experiment was conducted to provide a means to test the use of the hybrid FE model for aspect angles much greater than broadside. It was discovered that at certain aspect angles there exists a ray path that allows for multiple scattering between the cylinder and the interface, an effect that the hybrid model is not equipped to handle. This exercise not only illustrates the value in having models that can accurately capture the important physics driving a targets response, but also the importance of validating these models using controlled experiments.

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AN EFFICIENT NUMERICAL TARGET STRENGTH PREDICTION MODEL: VALIDATION AGAINST ANALYTIC SOLUTIONS

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Abstract: A decade ago, TNO developed RASP (Rapid Acoustic Signature Prediction), a numerical model for the prediction of the target strength of immersed underwater objects. The model is based on Kirchhoff diffraction theory. It is currently being improved to model refraction, angle dependent reflection and transmission (possibly through anisotropic materials) and multiple reflections and transmissions. These improvements are validated against available analytic solutions for simple shapes such as cylinders and spheres. Validation efforts will also be conducted against other numerical models using complex shapes such as a model submarine in the second Benchmarking of Target Strength Simulation (BeTSSi II) workshop that will be held in September 2014. The target strength model and a comparison of the numerical results with analytic solutions are presented. In addition, the effect and relative importance of the various improvements are evaluated.

Keywords: Target strength, numerical modelling, Kirchhoff diffraction, validation
1. INTRODUCTION

The target strength characterizes the strength of the response from a structure to emissions by active sonars, i.e., it determines the detectability of the structure by active sonars. It is a function of frequency, angle of incidence (and angle of observation for bistatic sonars), and of the size, geometry and materials of the (internal and external) structure.

Target Strength (TS) modelling and prediction can be used at the design stage of structures such as submarines. An accurate prediction model enables testing various designs in order to select the most suitable to meet specified TS-requirements. An efficient model allows for fast computation, thereby allowing more design iterations and/or reduction of the design phase duration.

The use of new materials, such as composite materials, influences the TS and raise the need for modelling capabilities to include such materials. Furthermore, as low frequency active sonars are becoming more widespread, there is an increased need for controlling and therefore modelling the target strength of submarines at low frequencies. The wavelengths at these frequencies are of the same order of magnitude as the characteristic dimensions of the structure, which imposes new challenges on modelling.

2. MODELLING

2.1. Kirchhoff diffraction theory

Kirchhoff diffraction theory is based on the Helmholtz-Kirchhoff integral theorem [1] that relates the amplitude of a wave field at any point inside a domain to its value and its normal derivative on the boundary enclosing the domain. Kirchhoff approximated this formula to study the diffraction of light passing through a slit in an opaque screen, assuming the considered receiver positions are in the far field of the slit. The formula is applicable to geometrical optics (when the wavelength is smaller than the slit) and enables analytic and numerical predictions. Although the theory was developed for light waves, it is equally applicable to acoustic waves.

2.2. RASP: Rapid Acoustic Signature Prediction

The same assumptions can be applied to scattering problems if the diffraction by (transmission through) a slit is replaced by the scattering by (reflection on) a rigid plane and will lead to a similar formula. Further approximations of Kirchhoff’s formula allow analytical evaluation of scattering by a rigid right-angled triangular (Fig.1a) element in the far field. As arbitrary triangles can be decomposed into two right-angled triangles (Fig.1b), the analytical formula enables evaluation of the far field pressure contribution of any triangle.

To model a structure, its geometry is decomposed into a triangular mesh and each triangle is further decomposed into two rectangular triangles. The pressure response of the structure is obtained by superposition of the contribution from each individual right-angled triangular element. The pressure of the scattered field is then used to compute the
target strength. For a simple structure such as a plate, the formula reproduces the specular reflection as well as side lobes (Fig1c).

Since its initial development around the year 2000, this general approach has been supplemented with two main improvements. The first is a reflection/transmission model to predict the response of elastic (non-rigid) structures. It is based on pressure reflection and transmission coefficients that model the local response of elastic materials. The pressure response of the triangle is modulated by the reflection/transmission coefficient corresponding to the material composition of the element. The reflection/transmission model includes dissipation and can simulate homogeneous materials as well as layered materials (including anisotropy, to some extent) and returns reflection/transmission coefficients as a function of the angle of incidence and frequency. The second improvement is a ray tracer that enables modelling of multiple reflections and transmissions, refraction and occlusion.

Fig.1: (a) Rectangular triangle local coordinate system. (b) Decomposition of a triangle into rectangular triangles. (c) Response of a triangle showing the main lobe in the specular direction and side-lobes.

3. VALIDATION FOR SIMPLE RIGID STRUCTURES

3.1. Analytic formula for a cylinder

The analytic formula for the monostatic target strength of a rigid cylinder of length $L$ and radius $a$ determined by Kerr [2] is:

$$TS_{cyl} = 10\log_{10}\left(\frac{aL^2}{2\lambda_{ref}^2}\left[\frac{\sin(kL\sin\theta)}{kL \sin\theta}\right]^2 \cos\theta\right)$$  \hspace{1cm} (1)
where $\lambda$ is the wavelength, $k$ is the wavenumber, $\theta$ is the incidence angle ($\theta = 0$ for normal incidence on the cylinder) and $r_{ref}$ is a reference distance equal to 1 m. In this model, the cylinder is rigid, infinitely thin and open at its edges. The derivation of the formula assumes that both the length and the radius of the cylinder are much larger than the wavelength, making this formula invalid at low frequency. Furthermore, the derivation assumes incidence angles close to normal incidence.

Note that the reference textbook on underwater acoustics by Urick [3] contains a table for the target strength of simple shapes (table 9.1) that includes the target strength of a finite length cylinder as described here. Although he cites Kerr, the formula is not properly reported as the cosine factor that appears is taken to the power 2.

### 3.2. Analytic formula for a sphere

Faran [4] determined an analytic formula for the scattering by a solid sphere immersed in a fluid. Unlike the formula for the cylinder, which relates only to the monostatic target strength, Faran’s solution represents the scattered fields in its entirety. It can be used to predict the monostatic and bistatic target strength of a sphere; it is valid at any frequency and for any type of homogeneous elastic material. For a rigid sphere of radius $a$, this analytic formula reproduces the well-known asymptotic results from Rayleigh [5]:

$$
TS(ka \ll 1) \approx 10 \log_{10} \left( \frac{25}{36} k^4 a^6 \right)
$$

$$
TS(ka \gg 1) \approx 10 \log_{10} \left( a^2 / 4 \right)
$$

### 3.3. Results

Fig.2 presents a comparison of the target strength determined using the analytic formula (1) and using RASP for a 40 m long, 4 m diameter rigid cylinder meshed using an element size of 0.5 m. At 1 kHz, which corresponds to a 1.5 m wavelength in water, the agreement between the two results is excellent, with discrepancies appearing only near $\theta = 90^\circ$, where the analytic formula ceases to be valid. Small differences are observed at 100 Hz even at normal incidence. However, since the wavelength is about twice the cylinder diameter, the assumptions underlying the analytic formula are no longer valid.

Fig.3 shows the target strength of the cylinder for two angles of incidence as a function of $ka$. For $ka \sim 0$, the two methods give results that are the same within 1 or 2 dB. At lower frequencies, the difference becomes larger, however since the analytic formula is known to be incorrect at lower frequencies, no conclusion can be drawn regarding the accuracy of RASP in this regime. At higher frequency however, the difference between the two methods decreases to a fraction of dB, until the vertical black lines after which the result from RASP diverges from the analytic solution. The divergence is due to geometrical meshing errors. These errors become significant when the difference between the surface of the mesh (made with flat triangular facets) and the real curved structure exceeds a fraction of the wavelength. The observed accuracy at high frequencies can be increased by refining the mesh. In the limit RASP converges to the analytic solution as both the mesh density and frequency are increased.
Fig. 2: Target strength of a 4 m radius, 40 m long rigid cylinder (mesh shown in inset): results from RASP and analytic solution at 100 and 1000 Hz.

Fig. 3: Target strength of a 4 m radius, 40 m long rigid cylinder: results from RASP (red) and analytic solution (blue) at 0° incidence (solid lines) and 25° incidence (dashed lines).

Fig. 4 shows normalized results for a rigid sphere. The analytic solution is shown in dark blue. At high \( ka \) (>15), its numerical implementation becomes unstable and gives incorrect results. The true solution is then approximated well by the asymptotic solution (green line at 0 dB). RASP was used to simulate the target strength of the sphere using two
meshes shown in the inset. The results using both meshes are very similar at low frequency. For $ka \ll 1$, the target strength is overestimated compared to the analytic solution. For $ka$ between 1 and 10, RASP response presents ripples that do not match the analytic prediction. In this regime, the maximum error is 6 dB. This error decreases with increasing $ka$ (it is less than 1 dB at $ka = 10$), until the mesh becomes too coarse for the considered wavelength. This happens just after $ka = 10$ for the coarse mesh (red line) and just after $ka = 40$ for the fine mesh (light blue).

Fig.4: Normalized monostatic target strength of a rigid sphere.

4. CONCLUSION AND FUTURE WORK

The RASP model is based on Kirchhoff diffraction theory and was adapted to simulate the response of complex elastic structures including multiple scattering. Comparison of the prediction of RASP for simple rigid structures indicates that RASP overestimates the TS at low frequency (which is acceptable as it enables conservative design), provides reasonably accurate prediction (within a few dB) in the mid frequency range ($ka$ between 1 and 10) and accurate prediction (a fraction of dB) at high frequency, providing the mesh is fine enough.

Further effort will be aimed at validating RASP using elastic structures with complex geometry. This will be performed by comparing RASP results against results of other analytical formula, finite element simulations and, within the BeTSSi II workshop [6], against other numerical models for TS prediction of submarines.

REFERENCES

Session 13

Habitat Mapping: Procedures and Results

Organizers: Philippe Blondel and Andrea Caiiti
TOWARDS JOINT USE OF SIDE SCAN SONAR AND SUB-BOTTOM PROFILER DATA FOR THE AUTOMATIC QUANTIFICATION OF MARINE HABITATS. CASE STUDY: LOURDAS GULF, KEFALONIA ISL., GREECE.

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Abstract: Although Multi-beam Echo Sounder tends to be the preferred tool nowadays towards gaining knowledge about the seafloor, numerous surveys have been and are still being performed by simultaneously using conventional Side-Scan Sonars (SSSs) and Sub-Bottom Profilers (SBPs). This is in respect of gaining fast, wide scale, three dimensional imaging of the seafloor and its substrate, extracting maximum value from a single and time limited survey. The combination of these two systems offer good knowledge of both the stratigraphy and the habitats of the seabed, aspects often linked to each other. However, a basic drawback is the inconsistency between their mapping scales, while SSS produces high resolution backscatter maps (of much higher density than MBES ones) while SBP produces substrata information of high vertical but very low horizontal density. In this work, 100 kHz SSS and 3.5 kHz SBP data, collected simultaneously during a geophysical survey at Lourdas gulf, Kefalonia Island, Greece, underwent post-processing and analysis, to extract numerous statistical features from both the seafloor and its substrate, towards automatic seafloor classification. The SSS records were mosaicked using Geocoder and the mosaic image was subjected to textural analysis, using the SonarClass software tool, to extract a large number of features. Unsupervised classification of the SSS features leaded to an accurate segmentation of the seafloor into homogenous regions. The SBP images were processed using multi-scale elongated steerable filters, to detect all seismic reflectors, and numerous features were extracted regarding the acoustic transparency and density of the seismic reflectors (layering) as well as the rugosity of the seabed. Supervised classification of the SBP features exhibited their high ability to discriminate between different known sea bed types, making them suitable for use as a substitute of traditional ground discrimination systems. The combination of the SBP track classes to the SSS segmentation, leaded to a full coverage - high detailed classification map of the study area.

Keywords: sidescan sonar, subbottom profiler, acoustic classification, habitat mapping
1. INTRODUCTION – STUDY AREA

Although Single Beam Echo Sounders (SBES) are traditionally the core for most automatic ground discrimination systems (AGDS), in this work, an effort to validate the high frequency Sub-Bottom Profiling (SBP) systems against revealing useful information about the seabed habitats in an automated way is realized. However, while AGDS use calibrated sounders, SBPs are inherently un-calibrated ones, and thus effort must be put to extract quantitative descriptors that are not significantly affected by variations in system operation setups, such as recording gains and transmitting power. Despite this complication, SBPs are able to reveal much more information about the seabed sediments and habitats than SBES, due to their penetration ability, and so the prospect of implementing a system to automatically quantify and classify SBP data seems appealing.

SBES and SBP methods have a great disadvantage comparing to the swath sonar systems, where data is collected from the most of the seabed, that they need to interpolation between the ship-tracks, thus less rigorously discriminate between defined seabed classes. Yet, information provided by swath and single beam sonar systems regard totally different physical properties of the sea-bed and they both deserve to be accounted to gain better insight into the seafloor processes and habitats. Through SSS images one can delineate and describe the different seabed types and features in terms of their acoustic intensity and texture. On the other hand, high frequency sub-bottom profiling systems offer information about the roughness and the hardness of the seabed, parameters strongly correlated to the sediment type and its vegetation. Concurrent substrata information may be used to quantify the acoustic transparency and layering of the superficial sediments and to detect the possible substrate, which, if it is close to the seafloor, it may influence the seabed biota.

In this work 3,5Khz SBP and 100Khz SSS data, collected simultaneously during a geophysical survey at Lourdas gulf, Kefalonia Island, Greece, are combined to produce full coverage classification maps. Lourdas exhibits a complex seafloor with frequent
changes in sediment and habitat types, making it a really challenging environment for automatic acoustic classification processes. During the coupled SSS - SBP survey, 94.2 km of way-lines have been carried out, insonifying a total area of 16.4 km² (fig.1.a). The validation of the seabed types was realized through a 7.2 km long ground truthing survey using a tow camera, which, in combination to the SSS mosaic and the SBP data, led to an expert manual classification, as shown in fig.1.b, some classes of which can be merged to produce a rougher classification scheme (see legend in fig.1.b). According to the suggested methodological scheme, the SSS mosaic is firstly used to segment the seafloor in large homogenous areas, while descriptors extracted from the SBP records are subjected to supervised classification trained by data samples in proximity to the groundtruthing points. Full coverage classification maps are finally produced by assigning its majority class to each one of the segmented areas.

2. SIDESCAN SONAR MOSAICKING AND SEGMENTATION

Acoustic returns from SSS produce intense geometric and radiometric artefacts in the created backscatter mosaics, which in turn affect the accuracy of facies delineation. Although Geocoder [1] has mainly been designed for use with MBES backscatter data, it can be used as well for raw SSS data, allowing the creation of more visually consistent mosaics. Specifically, application of its advanced AVG, despeckle and blending options leaded to a drastically improved SSS mosaic in Lourdas Gulf (fig.1.a). Backscatter values from different acquisition lines were reduced to a common scale of scattering strength, while their overlapping areas were blended so that only the best quality data is kept. The above, in conjunction with AVG application, leaded to a beautified mosaic.

![Validation index](image)

![Clustering (9 classes)](image)

Fig.2: Details from the two SSS segmentation scales: a. EDISON superpixels, b. fuzzy c-means clustering and c. Validation index suggesting 9 classes.

The segmentation of the mosaicked data was performed in two different scales. The first involved fine scale segmentation using the mean shift method (fig.2.a), as implemented in the EDISON software [2]. At this stage, the segmented regions, usually called superpixels, are quite small and so few of them coincide with the SBP way-lines. We need a larger scale segmentation, the separate regions of which will still honour the
exact boundaries of the various sea-bottom types, while most of them will coincide with a considerable number of SBP points. For this, we utilized clustering of the SSS mosaic via textural parameters extracted from 30x30m distinct image patches. Feature extraction and clustering took place through the Matlab tool SonarClass [3]. It employs three feature extraction algorithms, namely first order grey-level statistics, grey level co-occurrence matrices (GLCMs) and 2D power spectrum specifications, constituting feature vectors of totally 11 descriptors. The clustering took place using the fuzzy c-means algorithm and the optimal number of classes was decided upon the Calinski-Harabasz validation index, which clearly suggested 9 classes (fig.2.c). For each superpixel created in the fine scale segmentation stage, the median value from the underlying SonarClass classes was estimated, to produce a higher resolution classification map. This is moreover a noise-free, smoother version of the raw SonarClass classification map due to the local median consideration. What we need at this stage is the separate homogenous areas produced by the clustering process rather than their class assignments. This way the output is a segmentation map rather than a classification one.

3. AUTOMATIC TRACKING OF SEISMIC REFLECTORS

A set of local (pixel wise) contrasts \((c_{w,h,k})\) are defined, each extracted using couples of adjacent rectangular features of width \(w\) and height \(h\), rotated at direction \(k\) (see fig.1). Filtering the image using such rectangular features of a single direction, results in a directional boundary enhancement filter. In this work, that the enhancement of prolonged seismic reflectors is needed, we set the feature’s dimension to a fixed \(w\) to \(h\) ratio equal to 1:16. Thus, from now on, a certain scale \(a\) refers to a feature of dimensions \(w=a\) and \(h=a/16\) spatial units, giving the \(c_{a,k}\) local contrast. Contrasts are further normalized by taking their ratio to their maximum possible value, which normally is the range \((R)\) of data values (e.g. 256 in the case of 8bit images). In this way, local contrast takes values between 0 and 1, with unity implying the maximum contrast possible. According to the above, the normalized local contrast \((C_{a,k})\) for a fixed rotation \(k\) and scale \(a\), is defined as:

\[
C_{a,k} (x,y) = \frac{c_{a,k} (x,y)}{R} \tag{1}
\]

Rotation invariant normalized local contrast of a given scale \((a)\) is defined as the average \((\mu_{a,n})\) of local contrast along a set of \(n\) directions (eq. 2). This defines a rotational invariant boundary enhancement filter. An additional elementary statistical parameter derived in the context of this work is the standard deviation \((\sigma_{a,n})\) of the contrasts along the \(n\) directions (equation 3):

\[
\mu_{a,n} (x,y) = \frac{1}{n \cdot 6} \sum_{k=1}^{4} C_{a,k} (x,y), \quad \sigma_{a,n} (x,y) = \sqrt{\frac{1}{n-1} \left[ \sum_{k=1}^{4} C_{a,k} (x,y)^2 - \left( \sum_{k=1}^{4} C_{a,k} (x,y) \right)^2 \right]} \tag{2, 3}
\]

The above boundary enhancement filters are dependent on the scale under consideration, which can decrease their efficiency against reflectors of different thicknesses. For this, \(\mu_{a,n}\) and \(\sigma_{a,n}\) are averaged over a number of \(s\) successive scales, finally providing rotation and scale invariant boundary enhancement filters \(\mu_{s,n}, \sigma_{s,n}\) (eq. 4,5).
Another statistic measure considered, coefficient of variation: \( CV_{s,a} = \frac{\overline{\sigma}_{s,a}}{\overline{\mu}_{s,a}} \), proved to be beneficial for the accurate tracking of the seabed (see fig.3).

\[
\overline{\mu}_{s,n}(x,y) = \frac{1}{s} \sum_{a=a(1)}^{a(z)} \mu_{\alpha,a}(x,y), \quad \overline{\sigma}_{s,n}(x,y) = \frac{1}{s} \sum_{a=a(1)}^{a(z)} \sigma_{\alpha,a}(x,y)
\]

(4)

In the context of the present work, four feature scales were chosen in a quadratic fashion, namely \( a(1-4)=4, 8, 16 \) and 32m and four rotations, \( k(1-4)=0, 45, 90, 135^\circ \). In order to achieve maximum computational speed, the above filters were implemented using Haar-like features and integral images, as described in [4]. After successively filtering an SBP image, detection of the seismic reflectors was performed by finding the peaks in \( \overline{\mu}_{s,n} \) as a function of depth, for equidistant pings. An example of applying boundary enhancement filtering and seismic reflectors detection on a typical SBP image is shown in fig.3.

![Raw SBP image](image1)

![\( CV_{s,n} \)](image2)

![Detected reflectors](image3)

![\( \overline{\mu}_{s,n} \)](image4)

**Fig.3: A Raw SBP image, \( \overline{\mu}_{s,n}(x,y) \) \( CV_{s,n} \) filtering results, and the detected reflectors**

### 4. PARAMETERIZATION OF THE SEISMIC PROFILES

A total number of 28 features have been extracted from equidistant points (pingwise) of the seismic profiles (Table 1). Five of them regard the local bathymetric profile using rugosity and curvature definitions, eighteen use \( \overline{\mu}_{s,n} \) to quantify the distinctness and density of the seismic reflectors, three regard simple counting of the tracked reflectors and estimation of the maximum penetration ability into the sediments while finally two features concern the intensity change, in the raw SBP images, in the first few meters below the seafloor. Table 1 gives a short description for each extracted feature.
5. RANKING AND VALIDATION OF THE SBP FEATURES TOWARDS SUPERVISED CLASSIFICATION

The value of the designed parameters towards meaningfully quantifying the seabed is validated against their ability to produce accurate classification maps through supervised algorithms. Three classification algorithms, preceded by a feature selection wrapper method, were tested in WEKA software tool [5] against their performance about both the fine classification and the rough (merged classes) classification scheme. Two training sets were used, one consisting of a random 10% of the whole dataset (Exp.10%), labelled according to the expert classification map and another consisting of SBP samples that are in proximity to the groundtruthing points (GT training dataset). The performance of the classifiers was estimated using three different metrics, namely average true positive rate, Kappa statistic and the mean area under the ROC curves.
Naive Bayes classifier, although producing simpler decision boundaries, it outperformed the others when using the more realistic GT training dataset. Although in the ideal case where ground truthing stations are randomly distributed in the study area (Exp.10%), Bayes Network would perform perfectly, it failed in the GT case, probably due to over-fitting bias. The rough classification scheme, seemed to be more easily validated by all classifiers, but the ROC area was significantly high even for the fine classification scheme, when using Naive Bayes. Accordingly, Naive Bayes is considered a good classifier if trained by realistic groundtruthing sample distributions. The later was used to produce the final automatic classification map for our area (fig.4.a). This shows significant similarity to the expert classification map, even though a simple classifier was used, trained only by samples that were close to ground truthing stations.

Table 2: The performance of 3 classifiers, validated against two different set of classes (see fig.1.2) and training sets. Exp.10% : training set using the 10% of the whole dataset, GT: training samples collected from SBP points in proximity to the ground truthing.

<table>
<thead>
<tr>
<th>Classifier</th>
<th>No of Classes</th>
<th>Selected features</th>
<th>Training Set</th>
<th>TP Rate</th>
<th>Kappa statistic</th>
<th>ROC area</th>
</tr>
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<tr>
<td>Naive Bayes Fine classes</td>
<td>1,3,4,6,17,19,23,26,27,28 : 10</td>
<td></td>
<td>Exp.10%</td>
<td>0.746</td>
<td>0.6942</td>
<td>0.955</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>GT</td>
<td>0.542</td>
<td>0.4552</td>
<td>0.882</td>
</tr>
<tr>
<td></td>
<td>Raugh classes</td>
<td>1,2,4,7,17,18,26,27,28 : 9</td>
<td></td>
<td>Exp.10%</td>
<td>0.868</td>
<td>0.8058</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>GT</td>
<td>0.746</td>
<td>0.6322</td>
<td>0.948</td>
</tr>
<tr>
<td>Bayes Network Fine classes</td>
<td>1,2,3,4,5,7,11,12,17,21,23,25,26,27,28 : 15</td>
<td></td>
<td>Exp.10%</td>
<td>0.874</td>
<td>0.8475</td>
<td>0.987</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
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<td>0.248</td>
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<td>0.654</td>
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<tr>
<td></td>
<td>Raugh classes</td>
<td>1,2,3,4,7,10,16,20,21,23,24,28 : 11</td>
<td></td>
<td>Exp.10%</td>
<td>0.922</td>
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</tr>
<tr>
<td></td>
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<td></td>
<td>GT</td>
<td>0.342</td>
<td>0.1951</td>
<td>0.685</td>
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<tr>
<td>Decision Tree Fine classes</td>
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<td>Exp.10%</td>
<td>0.86</td>
<td>0.8304</td>
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<td>0.3466</td>
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<td>Raugh classes</td>
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<td>0.874</td>
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<tr>
<td></td>
<td></td>
<td></td>
<td>GT</td>
<td>0.744</td>
<td>0.6428</td>
<td>0.848</td>
</tr>
</tbody>
</table>

Fig.4: a. The final automatic classification map, produced using Naive Bayes classifier and b. the GT training dataset, compared to the expert manual one.
6. CONCLUSIONS

Through this work it was made evident that high frequency SBP systems, like 3,5 khz pingers and chirps, can be successively used for automatic ground discrimination purposes. The suggested SBP quantification methodology proved to be suitable for supervised classification, indicating that at least some of the extracted features are meaningful. The feature selection stage revealed some certain parameters that seem to have higher discrimination powers. Still, many other parameters played a role in producing more accurate classifications. Finally, using the segmentation of the SSS mosaic leaded to full coverage, consistent classification maps. In the future it is important to test the suggested method in datasets from other areas to validate its repeatability.

7. ACKNOWLEDGEMENTS

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AUTOMATIC CLASSIFICATION OF BEDFORMS USING PHASE DIFFERENCING BATHYMETRIC SONAR

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Abstract: This paper classifies bedforms and habitat using side scan images from a phase differencing bathymetric sonar. The study area is the inner shelf, between 3 and 15 m depth, of Barra da Lagoa – Moçambique beaches located on the northeast of Santa Catarina’s island, Brazil. The data was collected with an EdgeTech\textsuperscript{®} 4600 540 KHz interferometric system (phase differencing bathymetric sonar) which outputs side scan sonar images and swath bathymetry, proving images that are 3 and 4 times the width of the water depth. The data covered an area of approximately 12 km\textsuperscript{2} and was collected using the softwares Hypack\textsuperscript{®}2013 and Discover\textsuperscript{®}, and processed with SonarWiz\textsuperscript{®} and SonarClass\textsuperscript{®} for side scan and Hypack\textsuperscript{®} for bathymetry. The preliminary results show an inner shelf dominated by finer sediments, but containing 0.5 to 0.7m lower elevation patches of coarse grain rippled sediments, validated comparing automatic and manual classification of the images on the SonarClass\textsuperscript{®}, which uses textural parameters. The different bottom types were classified using SonarClass\textsuperscript{®} and also validated with ground-truthing station besides the bedforms in accordance of literature classification.

Keywords: Inner shelf, Bedforms, Habitat Classification, Interferometry.
INTRODUCTION

The inner continental shelf is dominated by waves, currents and tide. These processes have the capability to generate variations over the grain size and the bed morphology along and/or across this area (SHORT, 1999). Cachione et al. (1984) identified that ripple scour depressions occur globally on the inner continental shelf, which consists of rippled scour depressions composed of poorly sorted gravelly and coarse sand adjacent to a well-sorted fine sand sheets. Murray and Thieller (1994) recognized the coarse and fine sand as being part of larger bedforms features developed in heterogeneous sediments, and hence coined the term ‘sorted bedforms’.

In this study, the interferometric sonar EdgeTech 4600™ was used to provide side scan and bathymetry data which were used to classify the seafloor habitats. The mosaicked SSS records were classified, in both supervised and unsupervised manner, using the SonarClass® Matlab tool (FAKIRIS & PAPATHEODOROU 2003 and 2009) and the results were validated via sediments ground-truthing stations.

The study area is in a microtidal (0.4m to 1.2m), east coast swell environment, Barra da Lagoa and Moçambique headland bay beaches. They are 12 km long and limited by headlands, in north and south (Figure 1). The inner continental shelf is smooth sloped, dominated by waves and characterized by medium to fine sand, with the particles tending to be finer offshore. (SCHMIDT, 2010).

Figure 1: Survey area and sediment ground-truthing stations positions.

MATERIALS AND METHODS

The data consists of side scan images, bathymetry and ground-truthing stations. The classification is based on the side scan images as well as sediment samples.

Acoustics data were collected with an EdgeTech 4600™ 540 KHz bathymetric sonar, in depth ranging between 3 and 15 meters and line spacing between 3 to 4 times the water depth. It was used a DGPS/Heading Novatel FlexPack6 antenna and also a MRU SMC-108. The Novatel outputs uncertainties at the order of 0.15m horizontal and 0.25m vertical, using Omnistar XP corrections, and the MRU 0.03° RMS for pitch, roll and 5 cm or 5% for heave.
On SonarWiz®, the water column (bottom track) was extracted, the signal was adjusted using a time variation gain (TVG) and the bedforms were measured. The corrected sonar records were mosaicked and exported in GEO-TIFF format for use in SonarClass® for habitat classification. This software utilizes three feature extraction algorithms, namely first order grey-level statistics, descriptors extracted from grey level co-occurrence matrices (GLCMs) and 2D power spectrum specifications, constituting feature vectors of totally 11 descriptors (FAKIRIS & PAPATHEODOROU, 2009). More information can be found in Fakiris & Papatheodorou (2007 and 2009).

In order to validate this classification, sediment samples were collected by Van Veen grab sampler in places where different sea-bed textures were realized in the side scan data (Figure 1). Those samples were washed and dried to eliminate salt and humidity, and then on stereo microscope, on the same scale, pictures were taken in order to visualize the particle’s sizes, consisting in a qualitative analysis.

All features were classified in accordance to Society for Sedimentary Geology (SEPM) and Short (1999).

RESULTS
BATHYMETRY
Analysing the bathymetry (Figure 2), verified that the isobaths are parallel to the shoreline, with a shallower southern area associated to the headland. Along the central to north shore, there are a few bars where the wave breaking formed different bottom features during the survey. In the north, there is an island, almost in front of the headland, that causes a narrowing channel increasing the velocity of the current, generating larger bedforms.

![Figure 2: Bathymetry and bedforms examples. (A) Magnify of the north area with the larger bedforms, (B) Magnify of the centre area with bars, and (C) Bars and the bedforms associated.](image)

SIDE SCAN IMAGES
It is possible to detect differences in the intensity of the images (Figure 3), sometimes associated to the nadir zone or some noise originated from the ship
motions, in response to the side boat mounted pole, but mainly related to different sediment types.

![Figure 3: Mosaic of side scan images presenting the contrast difference between features. (A) e (B) are a magnify of the lower elevation brighter patches.](image)

Brighter patches are detected, aligned perpendicularly to the shoreline, most of them occur in the inner continental shelf from the centre to south, while random and smaller patches are detected over the north side. They are asymmetrical, long-crested rippled coarse sand areas that present a wave length of 0.70 to 1.2m and a height from 0.3 to 0.6m, with approximately 0.5-0.7m scour depressions (Figure 4A).

Over the darker area, the ripples found are asymmetrical, long-crested with wave length between 0.6 and 1m and height range from 0.20 to 0.4m (Figure 4B).

The bedforms in front of the north headland are asymmetrical, catenary, presenting lengths in the order of 6 to 12m and height between 0.8 to 1.6m (Figure 4C)

![Figure 4: (A) Rippled coarse sand; (B) Rippled fine sand; (C) Bedforms in front of the headland](image)

**SEABED SEDIMENT SAMPLES**

In this study the sediment was qualitative analyzed. Two different samples are shown (Figure 1); one represents the brighter areas related to the coarser particles (rag_2) and the other, the darker, corresponding to the finer particles (raf_1) (Figure 5). These results are in accordance to the literature (CACHIONE *et al.*, 1984; MURRAY and THIELLER, 1994; RAINEALT *et al.*, 2013).
Figure 5: Relation between particle’s size, image contrast and bedforms. Magnifications at the order of 20X.

HABITAT CLASSIFICATION
The supervised classification recognized both classes, namely the higher intensity patches of coarse grain rippled sediments and the other, which is the fine sediment with random occurrence of bedforms. The unsupervised classification recognized those features, validating the supervised one (Figure 6). So, comparing the seabed sediment samples and both classifications, it is possible to identify that the coarse particles has a different acoustic signature if compared to the fine sediments area.

Figure 6: (A) Unsupervised and (B) Supervised classification.

CONCLUSIONS
Based on the results it is possible to affirm that those brighter patches of lower elevation rippled coarse sand have a different acoustic signature when compared with the predominantly finer area, which was validated with the seabed sediments and also by literature. Those patches are all formed by bedforms, different from the predominant darkest area, which presents random bedforms. SonarClass® seems to be a good classifier because it didn’t recognize only both classes but it also classified in different classes the noise associated to the survey conditions and the water column extraction, making them easy to be excluded from the analysis.

ACKNOWLEDGEMENTS

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REFERENCES

ACOUSTIC MAPPING OF SUBMERGED MACROPHYTES IN SELECTED LAKES OF THE DRAWIEŃSKI NATIONAL PARK

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Abstract: Submerged macrophytes are among the key elements in freshwater ecosystems, and they are often considered to be good indicators of ecological status and water quality. However, traditional methods of their investigation are destructive, tedious, costly and time consuming. Hydroacoustic measurements of submerged macrophytes were performed in 4 lakes of the Drawieński National Park, NW Poland, using split beam echosounder Simrad EK60 with 200 kHz transducer. Data were analysed with Sonar 5 Pro, macrophytes module. Maps of macrophyte cover and height were produced for the lakes using GIS techniques. Although all lakes were clear-water ones, providing good conditions for plant growth, the macrophyte density, their occurrence depths and health condition differed greatly between the lakes. Simultaneously with acoustic survey, macrophyte grab samples (using a rake attached to rope) were taken, which allowed for classification of vegetation assemblies. Charophytes (different species: Chara tomentosa, Ch. filiformis, Ch. aspera, Nitella opaca, Nitellopsis obtusa) were easily distinguishable hydroacoustically from other plants, as Eurasian watermilfoil (Myriophyllum spicatum) or pondweeds (Potamogeton sp.). Since they were also occupying different depths, they can be separated and assessed independently from other types of vegetation. Many Charophyte species are considered as biological indicators of water quality, thus hydroacoustic monitoring of them can be used to assess ecological status of inland waters as required by the European Water Framework Directive, and to declare Special Areas of Conservation under the Habitats Directive. Such rapid and non-invasive monitoring of submerged vegetation becomes particularly important when facing huge differences in the community composition and plant cover due to climate change and interannual fluctuations.

Keywords: hydroacoustics, GIS, WFD, water quality, environmental protection
1. INTRODUCTION

Submerged macrophytes are among key elements in freshwater ecosystems, and they are often considered as indicators of a good water quality (1, 2, 3). For the proper managements of the water bodies information on the aquatic vegetation, their coverage, biomass and species composition is needed. Also Water Framework Directive (WFD) requires monitoring of all surface waters, based on biological quality elements, including macrophytes. In Europe there is adopted a “Guidance standard for the surveying macrophytes in lakes” (4, 5, 6) and in Poland standard procedure has been developed by Ciecierska (7). Unfortunately traditional methods of plant assessment are tedious, man power and time consuming, and provide mainly qualitative information which is difficult to quantify. Optical methods (from satellites) can provide quantitative spatial information, but they are dependent on the weather and water clarity. Hydroacoustic techniques can provide large-scale information on aquatic vegetation, which is free of the above mentioned disadvantages, however procedures are technically quite advanced and require qualified personnel. The first reports on hydroacoustic assessment of macrophytes were in late 80-ties and 90-ties (8,9,10,11), and since then the methods were improved and new possibilities developed so that now-days they are much easier to handle and even can be partly automated (www.biosonicsinc.com). In spite of several new publications in this field (12, 13, 14, 15, 16, 17), hydroacoustic methods are not commonly used for studies of macrophytes. It has been proved in cited articles that hydroacoustic methods of macrophyte assessment are quick, accurate and cost effective, and surely they deserve more dissemination and promotion. The aim of this paper was to show the advantages of using acoustics for mapping macrophytes spatial distribution as compare to traditional rake methods, mainly due to their high accuracy and additional information on plant heights structure. We also attempted to distinguish hydroacoustically between different assemblies of aquatic vegetation.

2. STUDY SITE AND METHODS

Drawieński National Park is situated in the north-western part of Poland, which is a fragment of the South Pomeranian Lakeland. There are 20 lakes in the Park, which are largely diversified by their trophic state, area, and depth. Four lakes, Czarne, Marta, Piaseczno Duże and Zdroje, most representative from the point view of aquatic vegetation, were chosen to perform hydroacoustic surveys of macrophytes. Measurements were done from a small boat moving along zig-zag transects with a constant speed, using SIMRAD EK60 split beam echo-sounder at frequency 200 kHz. Transducer was firmly attached to the boat and was directed vertically down. Pulse length was set to 0.1 msec., and repetition rate to as fast as possible in order to get maximum resolution. GPS was connected to the echo-sounder to enable geo-positioning of the data. Data were stored on a computer and analysis was performed in laboratory using dedicated to macrophytes module of specialized acoustic software Sonar 5. The output of the software includes the following information: maximum depth of macrophytes colonization, average areal and volume coverage, depth and height of plant for each ping. For the purpose of mapping, averages for plant coverage and height were done for every 100 pings. Maps were produced using Inverse Distance Weighted (IDW) interpolation method.
3. RESULTS AND DISCUSSION

The general information about the lakes and studied macrophyte parameters are summarized in Table 1.

<table>
<thead>
<tr>
<th>parameter</th>
<th>Czarne Lake</th>
<th>Marta Lake</th>
<th>Piaseczno Duże Lake</th>
<th>Zdroje Lake</th>
</tr>
</thead>
<tbody>
<tr>
<td>Secchi Disc* [m]</td>
<td>8.9</td>
<td>6.6</td>
<td>5.6</td>
<td>2.9</td>
</tr>
<tr>
<td>Average depth [m]</td>
<td>11.2</td>
<td>7.7</td>
<td>7.6</td>
<td>2.9</td>
</tr>
<tr>
<td>Max depth [m]</td>
<td>29.0</td>
<td>25.0</td>
<td>25.9</td>
<td>4.3</td>
</tr>
<tr>
<td>Plant coverage [%]</td>
<td>27</td>
<td>44</td>
<td>36</td>
<td>74</td>
</tr>
<tr>
<td>Max plant height [m]</td>
<td>1.8</td>
<td>2.8</td>
<td>2.2</td>
<td>3.0</td>
</tr>
<tr>
<td>Average plant height [m]</td>
<td>0.4</td>
<td>0.3</td>
<td>0.5</td>
<td>0.6</td>
</tr>
<tr>
<td>Max depth range [m]</td>
<td>10</td>
<td>10</td>
<td>10</td>
<td>4</td>
</tr>
</tbody>
</table>

* late summer readings at the time of field studies

*Table 1: General information about the lakes and studied macrophyte parameters.*

The preliminary results show that different vegetation assemblies can be classified in the echogram even “by eye”, at least basic groups such as Charophytes (different species: *Chara tomentosa*, *Ch. filiformis*, *Ch. aspera*, *Nitella opaca*, *Nitellopsis obtusa*) were easily distinguishable hydroacoustically from other plants, as Eurasian watermilfoil (*Myriophyllum spicatum*) or pondweeds (*Potamogeton sp.*, mostly *P. perfoliatus*). Fig. 1 presents typical example of Charophytes (Fig. 1a), pondweeds (Fig. 1b) and mixture of the two (Fig.1 c). It is common that these different plants not only look differently, but they also have different height and occupy different depths (Charophytes being distributed much deeper), which makes the differentiation more easy (Fig. 2).

The maps of average plant coverage and its height structure are presented in Fig. 3-6. They show high variability for the lakes studied mainly related to differences in depth and trophic status, both interrelated for these set of lakes.
Fig. 1 examples of different plant assemblages recorded in studied lakes:
a) charophytes, b) pondweeds and c) charophytes and pondweeds.

Fig. 2 Example of height structure of the vegetation from the Lake Czarne (echogram presented in Fig. 1c).
Fig. 3 Lake Czarne – from the left bathymetric map (source DPN) and maps of average plant coverage and macrophyte height.

Fig. 4 Lake Marta – from the left bathymetric map (source DPN) and maps of average plant coverage and macrophyte height.
Fig. 5 Lake Piaseczno Duże – from the left bathymetric map (source DPN) and maps of average plant coverage and macrophyte height

Fig. 6 Lake Zdroje – from the left bathymetric map (source DPN) and maps of average plant coverage and macrophyte height
Although lack or low macrophyte densities on deep, aphotic lake bottom areas are obvious, in many of our transects we also noted low macrophyte densities in lake sublittoral, at depths up to ca. 5 m, where light conditions should favor a dense submerged vegetation. This was different than indicated by our qualitative observations from the previous year. The difference was exceptionally long winter and sudden intensive warming of water and shallow sediments during a hot late spring in the year of studies. It is argued that enhanced – by high temperatures - sediment decomposition had led to lowering the redox potential of sediments in the shallower areas, thus reducing germination of plants and their early growth phases.

Such climatic anomalies may lead even to drastic changes in plant community composition, already observed there (18), or even to a shift toward another alternative state of the lake ecosystem (19). In this case it is often resulting in undesirable and durable dominance of Cyanobacteria or other bloom-forming planktonic algae. The application of acoustic monitoring in the early seasonal phase of vegetation development may become an efficient tool in indicating a probability of such a shift event, with subsequent appropriate counteracting measures undertaken, if feasible and desirable. Facing actual effects of climate change and uncertainty in ecosystem functions, a proper acoustic survey of submerged vegetation will be an important feature in characterizing the ecological status of vulnerable lake ecosystems, providing time- and cost-effective, adaptive guidelines for their conservation and management.

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REFERENCES


HIGH-RESOLUTION MULTIBEAM MAPPING OF HABITATS IN THE EXTREMELY SHALLOW WATERS OF THE VENICE LAGOON

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Abstract: Recent advances in multibeam technology have opened new possibilities in the field of mapping bottom morphologies, substrates and habitats. In particular, substrate mapping by means of backscatter and water column multibeam data is relatively new in extremely shallow environments, where multibeam systems have rarely been used. In this study we present the first results of an extensive survey carried out during 2013 in the very shallow channels of the Venice Lagoon using a Kongsberg EM-2040 DC multibeam system. Although numerous biological studies have been carried out to study habitats in the tidal flats and salt marshes of the Venice lagoon, the channel habitats are still almost unexplored. The new applications in habitat mapping given by the analysis of high-resolution bathymetry, backscatter and water column data will be discussed. In particular, we will show the first results of a survey carried out in a natural tidal channel of the northern part of the Venice Lagoon (water depths from 15 m to less than 1 m). On the high-resolution backscatter data collected, we carried out a two-dimensional (2D) textural analysis with the TexAn software. We then performed an unsupervised classification of the backscatter data. As a result, we were able to identify different backscatter areas where several grab samples were collected for ground truthing. With the help of this sampling, we calibrated the textural analyses and obtained a classification of the different kinds of substrate. The characterization of the substrate was then tested with a set of bottom photographs. In addition, the first results of combined bathymetric, backscatter and water column data analysis for bottom vegetation detection in the shallow lagoonal channels will be also shown.

Keywords: Habitat mapping, multibeam survey, algae detection, Venice lagoon

1. INTRODUCTION

In the last decade the advances in marine acoustic survey technique and, in particular, multibeam echo-sounder (MBES) technology have opened new frontiers in the field of mapping bottom morphologies, substrates and habitats (see [1] for a review). While these methods start to be widely used in shallow to deep waters (more than 10 m) (e.g.[2-5]), they were only very recently applied in extremely shallow tidal environments. This is due to the limitations brought on by water depths and the difficult environmental conditions ([6]). In fact, substrate mapping using backscatter and water column data is relatively new in extremely shallow environments, where acoustic methods have rarely been used (e.g.[7]). However, extremely shallow environments often undergo strong human pressure that needs to be constantly monitored. This is the case in the Venice Lagoon, the largest in the Mediterranean Sea (Fig. 1), where anthropogenic impact comes not only from the historical city of Venice, but also from a large industrial area (Marghera). The inlet is also undergoing major changes as barriers are built to defend the city from sea level rise. These
modifications of the inlets may alter or have already altered the lagoon’s benthic communities.

Fig. 1. General map of the Venice lagoon with the bathymetry and two zooms on the study areas: Canale Contorta-S. Angelo (bottom left) and Scanello (bottom right) channels

Many quantitative studies of the benthic community in the Venice Lagoon took place throughout the past century. However, channels habitats and biodiversity were rarely explored. The only extensive survey of these biocoenoses was carried out in 1930-1932 ([8]). It also was the very first benthic community study in the Lagoon. Subsequent studies and monitoring focused on the mudflat communities, with the exception of a couple of studies with a very limited spatial extent (e.g. [9]).

In 2013, within the National Flagship Project RITMARE, we carried out an extensive, high-resolution (up to 5 cm) MBES survey to explore all channels of the Lagoon, covering a total area of 50 km². During this survey, from April to December 2013, we acquired a large MBES dataset, including bathymetric, bottom and water column backscatter data.
The combination of MBES bathymetry and backscatter, together with ground validation samples, provides a robust means of mapping bottom substrates and benthic communities (i.e. oysters, algae, seagrass, etc.) ([10] and references therein). However, to produce such maps, it is very important to properly segment the MBES data into acoustic facies (i.e. areas with similar acoustic properties). Conventional, visual interpretation can be very effective if there are very distinctive features (e.g. rocks and bare, sandy bottom) but it can become very subjective where the seafloor is very heterogeneous or where its acoustic properties change gradually, with no clear separation from one backscatter area to another. To overcome this problem, we conducted an objective (semi-) automated classification of seafloor backscatter data (for an overview of different segmentation techniques see [2]).

The classification was done by comparing with sediment samples and photographs collected in the study area (the natural tidal channel in the Northern Lagoon, depicted in the bottom right of Fig. 1). Finally, we present some preliminary results of the analyses of MBES intensity water column data for underwater vegetation detection.

2. MATERIAL AND METHODS

2.1. Acoustic data

Multibeam data were acquired with a Kongsberg EM2040 Dual-Compact MBES during a survey carried out in April 2013. The MBES was on a pole on the 10-m long vessel Litus. The double-head MBES has 800 beams (400 per swath) and a frequency that can range from 200 to 400 kHz. During the survey, the frequency was set to 360 kHz and the data were acquired in equidistant mode, ensuring more than 30% of overlap between different survey lines. To have high-accuracy attitude corrections, we used the Seapath 300 positioning system with a Fugro DGPS correction. Sound velocity was measured continuously, close to the transducers, with a Valeport miniSVS sensor. Moreover sound velocity profiles were taken regularly with an AML Smart-X sound velocity profiler.

The tide corrections in all areas were computed with hydrodynamic model SHYFEM ([11]), giving the values of water level in 93 locations of the lagoon. The model computes sea level at each location (station) using wind and sea level data from all the tidal stations in the lagoon and at the inlets as forcing factors or as data to assimilate. The error of the model in the sea level simulation at a given station is of about 2 cm. All corrections are referred to the local datum Punta Salute 1897.

In order to obtain the digital terrain model files and the mosaics we used the CARIS Hips & Sips accounting for sound velocity variations, tides and basic quality control.

2.2. Textural analysis of backscatter data

In sonar imagery classification, texture refers to the distribution of acoustic energy and their positions relative to each other ([12],[13] and references therein). Textural analyses quantitatively describe the grey levels and their spatial relationships in small windows throughout an image. Grey Level Co-occurrence Matrices (GLCMs) have been shown to be the most adaptable tools for textural analyses of sonar data. Two textural indices, entropy and homogeneity, are sufficient to describe the GLCMs and resolve most textures visible in sonar imagery. Entropy measures the lack of spatial organisation inside the computation window, whereas homogeneity quantifies the amount of local similarities. Entropy and homogeneity indices were calculated for various values of the number of grey levels $N_G$ (from $N_G = 8$, by increasing powers of 2 until $N_G = 256$, corresponding to the full dynamic range of the sonar image), the window size (from $10 \times 10$ pixels to $80 \times 80$ pixels, by increasing steps of 10.
pixels) and the inter-pixel displacement $D$ (from 5 pixels within the computation window, to its maximum size minus 5 pixels, by increasing steps of 10 pixels). Maps of entropy and homogeneity were generated using these parameters. To ensure that the textural indices are not significantly influenced by the angle of ensonification, the co-occurrence matrices were averaged for angles of $0^\circ$, $45^\circ$, $90^\circ$ and $135^\circ$[12].

2.3 Ground-truthing

Ground-truth sampling (underwater photographic surveys and grab samples) was performed in order to interpret the seafloor features and characterise main subtidal habitats. Specific operative protocols were developed to cope with the main issues related to the distinctive features of lagoon channels: heterogeneous seafloor features, extremely high turbidity, strong tidal currents, and intense vessel traffic. The photographic surveys were performed on 22/11/2013 and 10/12/2013, around neap tides and slack water. Three 25-m transects (two of them almost overlapping) were positioned over the study area, in order to cover the main spatial units identified on bathymetric and backscatter data. They were arranged roughly along the direction of the current, over a range of depths (from -6 m to -1.7 m). Coordinates of the transect extremities were measured with DGPS. Pictures of 50-cm$^2$ photoquadrats were collected (moving upstream) every 5 m on both sides of the transects by professional technical divers (Nucleo Sommozzatori della Polizia di Stato di Venezia: State Police Divers). Each dive (transect— red lines in Fig. 3A) was done by a single diver, and lasted about 20 minutes. This kind of approach allowed the accurate georeferencing of the underwater images, necessary to compare them with high-resolution acoustic images. Moreover, eleven samples (with three replicas) were collected over the study area by a Van Veen grab (7 L). Sediment grain size (first 5 cm) was measured by laser diffraction (LISST 100X). Bioconstructors and, more generally, habitat formers, as well as other megabenthos, were evaluated by analyzing digital photos and sampled specimens.

3. ANALYSIS AND CLASSIFICATION OF SEA FLOOR BACKSCATTER

In the present study, we "ground truth" acoustic data to map seafloor habitats in the Scanello Channel (Fig 3). The specific protocol developed for the lagoon channels proved to be effective. Photographic surveys allowed characterising a number of habitats in terms of physical structure and main biota. Backscatter patterns can be referred to different seafloor features, including sediment textures and hardness, as well as cover. Depositional and erosional landforms coexist over a limited spatial extent. Recognised substrate types include bare sediments ranging from coarse silt to very fine sand (see for example the bottom photographs in the bottom right corner of Fig. 3A), reflecting the role of currents in shaping the channel structure. Depending on the local conditions, they are sometimes characterised by coarse shell debris. In these units, photoquadrats shows almost 0% of phyto- or zoobenthos coverage, with the exception of few individuals of the alien bivalve *Scapharcaina equivalvis* (Bruguière, 1789) and of dog-welks (*Gastropoda: Nassariidae*), the presence of a benthic diatoms layer and some burrows probably attributable to the decapod *Upogebia* sp. This type of substrate was classified in dark red in Fig. 3B. This bottom type is associated to the lowest values of entropy and homogeneity obtained with the TexAn image analysis (Fig. 3C and D).
Complex biogenic structures developing on hard substrata extend over a large part of the study area, and can be easily recognised on backscatter data (see the upper left corner of Fig. 3A). The bioocoenosis mostly develops on dead oyster beds (*Crassostrea gigas* (Thunberg, 1793)). Oyster valves are encrusted and cemented by other bioconstructors such as tubicolous polychaetes, and colonised by habitat-forming species such as sponges and bryozoans. Main Demospongiae producing massive and spatially structured encrustations include *Haliclona* sp., *Halichondria* sp., *Dysidea* sp., *Tedania* sp. Cnidaria are represented by *Anemonia viridis* Risso, 1826. Brown and red algae such as *Gracilaria* are also present with extensive coverage, and are in turn characterised by epiphytic Bryozoans. These communities are
characterised by high biomass and relatively high diversity. This kind of benthic community was manually segmented within the blue class in Fig. 1B and it has the highest values of entropy and homogeneity (Fig. 3C and D). We then defined a third class (in yellow in figure 3B) of very coarse silt and very fine sand with shell fragments and scattered sponges agglomerate, that has intermediate values of entropy and homogeneity (Fig. 3C and D). The channel is characterised by strong tidal currents (up to 0.6 m/s) and high water exchange, also during neap tides. These are major factors in shaping these communities. A high amount of suspended solids, including particulate organic matter, is transported by currents, mainly during ebb tides. This suspended material causes high turbidity but supports as organic source suspension feeders, such as Bivalvia, Porifera and Bryozoa. At the same time, currents contribute to the export of catabolites during ebb tide and import of oxygen during flood tide. These processes reduce the natural saprobity of the system, which is typical of transitional ecosystems such as lagoons, consequently enhancing biodiversity ([14]).

4. VEGETATION DETECTION USING WATER COLUMN BACKSCATTER

Seaweeds are known to be acoustically visible ([15], [16]) and might be detected by MBES within different angular ranges depending on MBES technology, due to signal to noise ratio (SNR) [16]. We found out that, with the Kongsberg 2040DC MBES, large plants can be detected in all beams up to 70°. This is due to a very good SNR (Fig. 4) in all beams. Moreover, in the shallow environment of the Venice Lagoon, the dense bottom vegetation traps gas bubbles, which significantly increasere reflectivity. This phenomenon can also influence the accuracy of bathymetric measurements, creating false bottom detection in the water column (Fig. 5).

Fig. 4. Left: intensity values in water column data from one swath from two overlapping sonar heads (port: 340 kHz, starboard: 380 kHz). Right: two signals from beams -62° and +62° showing backscattering values for bottom, plants and sidetone noise.

The nominal frequencies of port (340 kHz) and starboard (380 kHz) heads differ by 40 kHz, influencing the intensity of the recorded signals. Bottom backscattering (BS) coefficients from the Kongsberg 2040-DC are corrected for this difference, the same as for angular dependence, spreading and absorption loss but not for scattering area and bottom slope ([17]). Figure 4 shows raw water column data for one of the swaths from an area in Fig. 5 and a comparison of intensities from two beams taken by different heads but at the same angle (62°). Plants are present in the port beam (black) between 42 and 50 sample and have higher backscattering values than sidelobes noise (25:42 samples).
4. CONCLUSIONS

Two-dimensional (2D) textural analysis of high-resolution backscatter data collected was carried out with the TexAn software. Thanks to the correlation with sediment samples and bottom photographs, backscatter patterns could be related to different sediment textures and covers. Recognised substrate types include bare sediments, ranging from coarse silt to very fine sand, macroalgae cover, and complex biogenic structures such as dead oyster beds. The latter are intensively encrusted by bioconstructors such as tubicolous polychaetes, and colonised by other organisms such as sponges, tunicates and bryozoans. They are characterised by high biomass and diversity. Arguably, the strong tidal currents and high water exchange are a major factor in shaping these communities.

This is completed with the first acoustic identification of macroalgae and seagrass in the Venice Lagoon, using MBES water column backscatter data, showing it can be very effective to map macroalgae and seagrass in tidal channel environments.

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DEVELOPMENT OF ACOUSTIC COLOUR TECHNIQUE USING MULTI-FREQUENCY SWATH ACOUSTIC BACKSCATTER

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\textbf{Abstract:} Swath acoustic data collected by multibeam sonar systems is a recognised tool to efficiently map the bathymetry and texture of large areas of seabed so aiding the definition of benthic habitats. Backscatter data may be spatially classified based on either standard measurements of characteristic acoustic angular response curves (ARCs) or backscatter model parameters (inverted from the ARCs) that depend on seabed physical properties. However, the inverted model parameter of “roughness” is intrinsically linked to the acoustic wavelength; only the roughness spectrum with wavelength less than half of the acoustic wavelength affects the surficial acoustic backscatter. Therefore, inferred surficial texture is intrinsically acoustic frequency-dependent. This research tests the feasibility of using multi-frequency acoustic backscatter to create acoustic classes based on a broadband acoustic response to natural roughness spectra. The goal is to generate more accurate spatial delineation of facies.

\textbf{Keywords:} acoustic classification, multibeam, multi-frequency

\textbf{EM2040D} (300 kHz (continuous wave (CW) pulse)), \textbf{EM710} (100 kHz (CW)) and \textbf{EM302} (30 kHz (CW)) benthic acoustic data were collected simultaneously using Kongsberg Maritime’s test vessel, Simrad Echo, within the Ormø–Færder Marine Protected Area in Oslofjorden, Norway. 47 line-km of multi-frequency data were collected resulting in a survey extending 2.9 km by 3.2 km. In addition to the latter overlapping survey lines, 5 non-overlapping, 2.9 km-long cross-lines were acquired to investigate anisotropic backscatter effects; the cross-lines were run perpendicular to bathymetric contours to minimise variation of possible depth-dependent benthic texture within a swath. This paper discusses the results of delineations and compares maps for datasets separately and jointly classified with different frequencies.
1. INTRODUCTION

As a mapping tool, multibeam echosounders (MBESs) are used primarily as bathymetric measurement devices. While the amplitude and angular dependence of the backscattered signal, from which the “sounding” is extracted, has long been recognised as a useful indicator of seabed rugosity, and by empirical extension, grain size [1], it is usually seen as being secondary in importance to the primary bathymetric function of MBESs. For vessels equipped with only one (narrowband) MBES, the subordinate ranking of multibeam backscatter data has no practical effect but for vessels equipped with more than one multibeam (or a single broadband MBES as suggested by [2]) where, for the dual purposes of highest bathymetric resolution and minimisation of data storage, usually only the highest resolution multibeam is in operation. Backscatter data from the lower frequency MBES is therefore not collected and, since the dominant seabed roughness, usually parameterised as the roughness spectral strength, $w_2$, and spectral exponent, $\gamma_2$, inferred from the angular dependence of MBES backscatter, is inherently acoustic wavelength dependent [3], we propose that potential roughness information is being lost by this practice. The inclusion of additional frequencies of acoustic backscatter, where possible, has been proposed as having great potential for more accurate seabed classification [2, 4, 5].

The assumption is usually made that the roughness spectrum conforms to a single power law and that $\gamma_2$ is a constant (usually -3.0 to -3.5 [1]); this allows inversion for $w_2$ and, by extension, grain size, to be MBES-frequency independent. However, lander-based studies of seafloor roughness spectra observe that the line of best-fit may be composed of at least two, and up to five, line segments [6] and that acoustic data either side of 150 kHz may be scattered from power laws conforming to different spectral parameters [7]. In addition to the latter theoretical basis, a relationship between frequency-dependent backscatter and sediment type has also been detected in MBES data from the Bay of Fundy giving rise to a suggested utility of multi-frequency MBES backscatter for seabed determination and classification [4]. Another feasibility study took a composite image-based approach to the classification of a small dataset of EM1002, EM3002 and Reson 7125 data in Galway Bay and showed promising results for the distinction between shell hash and maërl using the ISODATA algorithm [8]. There are practical issues for the implementation of multi-frequency MBES seafloor classification, namely the attenuation of high-frequency sound in deeper water and the increased size of lower frequency MBES rendering them impractical for survey launches [4], but setting these aside, this paper will examine the utility of multi-frequency MBES backscatter for more accurate seabed classification.

Kongsberg Maritime’s test vessel, Simrad Echo, was equipped with an EM2040S, EM710, and EM302 for the survey, carried out over two days within Ormø–Færder Marine Protected Area in Oslofjorden (Fig. 1). MBES acquisition parameters (Table 1), e.g. pulse length, were set manually during data acquisition to eliminate backscatter magnitude changes due to automatic depth-dependent pulse length adjustment [4]. Although the MBESs used are “next generation” echosounders, e.g. capable of transmitting chirped frequency-modulated (FM) pulses [2], they were operated in narrowband, continuous-wave (CW) mode to eliminate any possible backscatter artefacts arising from a broadband acoustic signal. The area was surveyed NW-SE with 100% bottom coverage and then re-surveyed with lines on a perpendicular heading. Under the assumption that seabed type was correlated with slope, the latter lines were run perpendicular to bathymetric contours in order that the assumption of homogeneous seabed within a “patch” of 30 consecutive half-swaths was upheld. Only
backscatter data from the slope-perpendicular NE-SW survey lines will be discussed in this paper.

![Image](image.jpg)

Fig. 1 Five-metre DEM of EM2040 data showing Roche Moutonée, precipitous slopes, and glaciated valleys, topographic features typical of the region.

<table>
<thead>
<tr>
<th>MBES</th>
<th>Frequency, [kHz]</th>
<th># Soundings</th>
<th>Transmit x Receive Beamwidth, [°]</th>
<th>Pulse Length (CW), [μs]</th>
</tr>
</thead>
<tbody>
<tr>
<td>EM2040</td>
<td>275, 290, 280 (3 sectors)</td>
<td>400</td>
<td>0.5 x 1.0</td>
<td>600</td>
</tr>
<tr>
<td>EM710</td>
<td>71, 83, 77 (3 sectors)</td>
<td>400</td>
<td>0.5 x 1.0</td>
<td>500</td>
</tr>
<tr>
<td>EM302</td>
<td>26.5, 30.5, 33.5, 28.5 (4 sectors)</td>
<td>432</td>
<td>2.0 x 2.0</td>
<td>750</td>
</tr>
</tbody>
</table>

Table 1: MBES acquisition parameters.

2. METHODOLOGY

A patch-based approach, whereby angular response curves (ARCs) are averaged in consecutive patches of 30 pings, similar to the approach of Fonseca and Calder [9], was taken with a further modification for spatial joining of co-located patch centroids from the three different frequencies. Using the research version of Geocoder [10], patch-averaged ARCs are decomposed into 5 descriptive parameters: gradient (“x_avo”) and intercept (at 10° incidence angle) (“y_avo”) of best-fit line to the ARC within the near range (0° to 25°); average AR within the far range (25° to 40°) (“f_avo”); average AR within the near range (“n_avo”); and the fluid factor that is correlated with volume heterogeneities (“ff_avo”) [11]. These patch-
based parameters are used as feature vectors for k-means classification but, since the patches corresponding to the different frequency MBES’s are independently derived and not co-located, they must be spatially “joined” using conventional GIS software. The Spatial Join process merges the three attribute tables containing the feature vectors for the individual MBESs (Fig. 2 (LEFT)) into one attribute table (Fig. 2 (RIGHT)) so that they can be statistically analysed for clustering and Principal Component Analysis. In practice, the attribute tables of the closest EM710 and EM2040 patch centroids were joined to the EM302 attribute table.

![Patch Centroids](image1)

**Fig. 2**: Port and starboard patch centroids of the different MBESs (LEFT) and (RIGHT) the spatial distribution of the JOIN-ed patch centroids.

K-means cluster analysis and Principal Component Analysis of feature vector attributes were carried out using a Clustering Toolbox [12]. To investigate the effect of different acoustic frequencies on the classification result, the frequency-related feature vectors were classified separately, in pairs (EM2040+EM710; EM2040+EM302; and EM710+EM302), and all together (EM2040+EM710+EM302) to give seven different classification results.

3. RESULTS

3.1. Backscatter Mosaics

![Backscatter Mosaics](image2)

**Fig. 3**: Backscatter mosaics of the MBES cross-lines discussed in this paper.

Time series backscatter data corrections [13] were applied in Geocoder as follows: removal of Lambert’s Law correction implemented by the MBES hardware during
acquisition, which assumes a flat seafloor; correction for pulse length dependent footprint area and correction for the actual incident angle using the DEM (Fig. 1). Backscatter data was then mosaicked (Fig. 3).

3.2. K-means Clustering and Principal Components Analysis

Principal Components Analysis (PCA) is suited to testing the discriminatory ability of classifications involving more than one acoustic frequency; ideally, in addition to expected correlation of ARC measurements of the separate ARCs, each frequency will contribute some degree of classification variation along independent orthogonal (“PC”) axes. The results of the k-means clustering are displayed in Principal Component space (Fig. 4), where PC1 and PC2, the primary and secondary principal components, are linear combinations of the feature vector attributes (x_avo, y_avo, f_avo, n_avo, and ff_avo) of the patch centroids of one or more of the MBESs used in this study. The cluster centres for classifications including the EM2040 feature vectors (Fig. 4(a, b, c, and g) exhibit a preferred elongation along the PC1 axis and the Eigenvalue of the second principal component is less significant compared to the classifications of the EM710 and EM302 data. This indicates that covariation in the EM2040 and EM710 feature vectors explains the majority of clustering and there is minimal orthogonally distributed, i.e. PC2-orientated data.

Fig. 4: K-means cluster centres (red crosses) displayed in Principal Component space with contours of Cartesian distance between data point and nearest cluster centre. Eigenvalues, indicating the significance of the component, are bracketed after the axis title.
The main descriptive ARC parameters for the first principal component of the EM2040 are the backscatter value at 10° incident angle, the mean backscatter values in the near and far angular ranges, and the Fluid Factor (Fig. 5(a)). The latter is also true for the EM710 with the addition of a weak dependence on the variation of the slope of the ARC in the nadir region (Fig. 5(b)). By contrast the EM302 exhibits is less discriminatory ability, with classes distributed more isotropically in principal component space (Fig. 4(f)) and the first two principal components are less distinct with eigenvalues of 0.5 and 0.4 respectively.

The power of combining the separate discriminatory characteristics of the EM2040 and EM710 is evidenced by an additional degree of classification freedom for the joint EM2040+EM710 classification (Fig. 6). While the EM710 ARC measurements clearly contribute to the first principal component of variation, indicating unsurprising correlation between the EM2040 and EM710 angular responses, the second principal component is notably strongly dependent on the EM710 ARC measurements alone (red bars, Fig. 6). This indicates that the EM710 is contributing additional discriminatory ability to the joint EM2040+EM710 classification.
3.3. Classification Maps

The results of the k-means clustering were mapped geographically using Voronoi polygons centred on the patch centroids (Fig. 7(g)). Assuming that the sediment types are in reality contiguous and correlated with bathymetry and/or slope, it can be seen that, in the absence of any other information, the classifications involving the EM710 are the most plausible, i.e. EM710 alone (Fig. 7(d)); EM2040+EM710 (Fig. 7(b)); and EM2040+EM710+EM302 (Fig. 7(g)). The classification of EM710 alone is quite similar to the classification utilising all three MBES ARC parameters but the latter displays an arguably more realistic distribution of classes four and five and rocky classes one and three are spatially more distinct (Fig. 8). However, the latter remark requires testing by detailed grab sampling, something not carried out to date.

Fig. 7(f) shows that the EM302 is a poor discriminator of the fine sediment within the deep, bifurcating submerged glaciated valley and the fine sediment on the broad flank to the west. However, the addition of EM302 ARC information to the joint classification of EM2040+EM710 Fig. 7(b) leads to better classification of the rocky substrate in the southwestern strait Fig. 7(g) and more plausible classification of class five in the southern extremity of Fig. 7(g).

Therefore the classification based on the ARC parameters of all three MBESs may be viewed as a “fine-tuning” of the EM710 (alone) classification.
Fig. 7 Classifications based on k-means clustering of ARC measured parameters for: individual MBES (Black outline); jointly classified MBESs (blue); and all three MBESs (red). Bathymetric contours are displayed in metres and patch centroid locations symbolised in (g)
4. CONCLUSION

This study has tested the feasibility of utilising backscatter from different frequency MBESs for more accurate seabed classification has shown that:

- In the absence of ground-truthing and assuming seafloor type is locally homogenous and related to bathymetry and/or slope, the classification using the EM710 alone produces a plausible regional seafloor configuration
- The inclusion of ARC information from EM2040 and EM302 may be viewed as a “fine-tuning” of the EM710 classification with two more distinctive rocky classes, that, we speculate, may be distinguished by thickness of sediment cover, and better definition of Class 5 sediment in the southern extremity of the field area.

Future work will utilise the entire dataset, i.e. inclusive of the NW-SE survey lines, investigate the use of more ARC descriptive parameters, and utilise other clustering approaches.

REFERENCES

MARTA: AN AUV FOR UNDERWATER CULTURAL HERITAGE

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Abstract: MARTA, acronym for MArine Robotic Tool for Archaeology is a small-sized Autonomous Underwater Vehicle (AUV) developed in the framework of the ARROWS project. The ARROWS project (start September 2012, end August 2015) is funded by the European Commission in the framework of the FP7 call ENV-2012, challenge 6.2-6, devoted to Development of advanced technologies and tools for mapping, diagnosing, excavating, and securing underwater and coastal archaeological sites. MARTA will operate in a heterogeneous team of vehicles with a common mission to perform and a distributed and shared world model updated based on non-synchronous information collected by each of the vehicles of the team. Each of the vehicles will be equipped with acoustic communication means in order to be able to communicate when submerged. The University of Florence is in charge for the design and construction of the MARTA AUV, according to specifications written in compliance with the requirements formulated by the Archaeological Advisory Group, including archaeologists both from inside and outside the ARROWS consortium. MARTA will operate at a maximum depth of 150m and, in addition to a pair of acoustic modems for inter-vehicular communication and USBL localization, it features two different payloads, i.e.: a pair of synchronised digital TV cameras with visible light as well as structured light (blue laser) illuminators; a Multibeam echo-sounder. The paper illustrates the vehicle concept and the use of on-board acoustic instrumentation for communication, localization, and sea-bottom imaging. Preliminary experimental data from the field are presented.

Keywords: Underwater Robotics, Underwater Acoustic Measurements, Autonomous Underwater Vehicle, Underwater Cultural Heritage
1. INTRODUCTION

The ARROWS project challenge is to provide the underwater archaeologists' with technological tools for cost affordable campaigns. Several technologies, originally developed for military use and the Oil&Gas industry, have been successfully adapted to underwater archaeology (e.g. acoustic communication or sub bottom profiling). However, the cost of underwater missions involving a surface ship and Remotely Operated Vehicle (ROV) is in excess of 50000€/day; therefore, there is a strong motivation for archaeologists to reduce the costs associated with underwater campaigns otherwise impossible to perform without the support of private sponsors and/or foundations.

ARROWS project is funded by the European Commission in the framework of the FP7 call ENV-2012, challenge 6.2-6. The project is coordinated by the University of Florence (IT) and its consortium is composed of several research institutions and companies dealing with Underwater Robotics: CNR-ISTI (IT), Tallinn University of Technology (EE), Heriot-Watt University (UK), Edgelabs s.r.l. (IT), Albatros Marine Technologies (ES), Nesne Elektornik (TR), TWI (UK), Soprintendenza del Mare – Regione Sicilia (IT), Estonian Maritime Museum (EE). The ARROWS Steering Board is supported by the Archaeological Advisory Group, composed of European archaeologists whose task is to guide and follow the strategic developments of the project.

ARROWS adapts and develops low cost autonomous underwater vehicle technologies to reduce significantly the cost of archaeological operations, covering the full extent of archaeological campaign. The project aims to deal with underwater mapping, diagnosis and cleaning tasks. ARROWS formalizes the archaeologists’ requirements in all the phases of the campaign, identifies problems and proposes technological solutions with technological readiness levels that predict their maturation for exploitation within 3-5 years. ARROWS also addresses the issue of training archaeologists to the use of new equipment and techniques.

The systems and methodologies developed within ARROWS comply with the “Annex” of the 2001 UNESCO Convention for the protection of Underwater Cultural Heritage (UCH). The system effectiveness will be demonstrated in two places, different as regards the environment and the historical context, the Mediterranean Sea (Egadi Islands) and the Baltic Sea.

ARROWS is dealing with the development of a team of new heterogeneous AUVs to support archaeologists in all the phases (mapping, diagnosing, cleaning, and monitoring) of underwater campaigns. The components of the system will be easily deployable by a team of archaeologists during a mission with limited support by technicians. The archaeologists will be trained to use the innovative tools produced in the framework of the ARROWS project. Three classes of new AUVs are springing up according to archaeologists’ needs.

Innovative AUVs, developed in the framework of ARROWS, are (Fig. 1):

- MARTA AUV: MArine Robotic Tool for Archaeology – modular AUV, easily adaptable to the various types of mission according to its configuration;
- U-CAT – small biomimetic AUV, usable for shipwreck penetration;
- A-sized AUV – small torpedo-shaped vehicle, easily manageable thanks to its reduced size.
After a brief description of the general architecture of ARROWS team of AUVs (Section 2), the paper focuses on MARTA vehicle description (Section 3) and the use of its on-board acoustic instrumentation for communication, localization, and sea-bottom imaging. MARTA is a low cost vehicle, compared to commercially available AUVs, and light enough to be deployable by two people from a small boat. The maximum operating depth is equal to about 150 m; concerning its payload for patrolling and mapping, MARTA mounts a pair of synchronised digital TV cameras with visible light as well as structured light (blue laser) illuminators and a Multibeam echo-sounder. Finally, preliminary experimental data from the field are presented (Section 4).

2. GENERAL ARCHITECTURE OF THE TEAM OF VEHICLES

ARROWS project proposes a team of heterogeneous autonomous underwater vehicles capable of satisfying all the needs of a complete archaeological campaign. The differences among the AUVs involved in ARROWS are related to the different roles that have to be covered within the cooperating team. In particular, three main roles have been identified:

- **Search AUV**: AUV equipped with acoustic sensors such as Side-Scan Sonar (SSS) or Multibeam Echo Sounder (MBES) to be used for fast and large surveys, searching for points of interests (candidate points);
- **Inspection AUV**: the role of the inspection AUV is to reach the points, identified as potentially interesting thanks to the data acquired by the Search AUV, and to acquire optical
and/or acoustic images in order to confirm or not the candidate points and to obtain more
details;

- **Biomimetic Robot (BR):** its main role is the wreck penetration to get images from hardly
  accessible areas; the main features of BRs are small sizes and high manoeuvrability in
  addition to low cost, considering the not negligible risk of loss.

All the vehicles cooperate thanks to the distributed High-Level Control System (under
development at the Heriot-Watt University) that exploits a distributed world model built on
board each vehicle through the acquired knowledge and information received by the other
vehicles. This way, by exchanging only few synthetic information (to not saturate the
communication band), the AUVs will be able to conduct a common mission with a common
goal. Because of their simplicity and because the target environment (inside shipwrecks) is not
the same of the AUVs (outside the wrecks), the world model will not be implemented on the
BRs (U-CAT); this allows also a reduction of the required performance and, thus, of costs, for
their on board pc.

According to the ARROWS Archaeological Advisory Group (AAG) requirements, stated in
the first phase of the project, one of the main design criteria for MARTA is the modularity.
Depending on the mission to be performed, MARTA is designed to be configured with several
payloads and different propulsion systems.

This way, MARTA can play both the role of search and inspection AUV. In the framework of
the ARROWS project, the payloads on MARTA will be of acoustic and optical type, in
particular:

- **Acoustic payload:** a Multibeam Echo Sounder (MBES) can be mounted in the bow of
  MARTA; a bow module for the integration of a Teledyne BlueView M900 has been
designed and will be soon built;

- **Optical payload:** a section of MARTA houses the optical payload devices (a couple of
  Basler Ace cameras, a C-laser Fan from Ocean Tools, four illuminators produced at the
  University of Florence, a SBC Commell LS-378 for data acquisition and processing).

As concerns the underwater communication means, MARTA will be equipped with two
different acoustic modems:

- **Evologics 18/34:** it will be used for communication between the conventional AUVs
  of the team, allowing a high data transfer rate (13.9 kbit/s) and a high functioning range
  (3500 m); it will be used along with an USBL transducer by Evologics for localization
  and navigation purposes;

- **AppliCon SeaModem:** it will be used for communication with the BRs; U-CAT role
  justifies the use of a different communication means, characterized by a considerably
  lower cost although with lower performances (data transmission rate and operating
distance).

Thanks to this configuration, MARTA will represent a bridge node in the communication
between the AUVs and the BRs. An acoustic communication and localization protocol,
oriented to the optimization of the exchanging data rate and of the accuracy in sensory data
georeferencing, will be developed, tested and demonstrated during the final experimentation
of ARROWS. The protocol will take into account the constraints in terms of acoustic channel
sharing among a variable number of different acoustic nodes (Evologics USBL, Evologics
modem, AppliCon SeaModem). A TDMA (Time Division Multiple Access) protocol will be
the first solution; then an iterative procedure will bring to the definitive version according to the feedback coming from the simulation results and the testing activity outcomes.

3. MARTA AUV

MARTA prototype will be easily deployable from a small boat; the vehicle is modular and has a total length of about 3 m (as concerns its longest configuration – including the Optical Payload), an external diameter of 7 inches and an in-air weight of about 70 kg. The vehicle has 5 degrees of freedom fully controllable by means of 6 actuators (electrical motors + propellers): 2 rear propellers, 2 lateral thrusters and 2 vertical thrusters. In Fig. 2, the final 3D CAD of MARTA AUV is shown.

![Fig. 2 – MARTA final CAD](image)

In summary, the vehicle characteristics are:

- Reachable depth: 150 m;
- Vehicle longitudinal speed: maximum reachable speed of 4 knots;
- Autonomy: about 4 hours;
- Dimensions: length of about 3 m, external diameter equal to 7 inches;
- Weight: about 70 kg;
- Power supply - voltage: 24 V;
- Modularity: the archaeologists involved in ARROWS looked for specialized vehicles, i.e. suitable AUVs with specific sensors for specific missions. MARTA is thus configurable thanks to its different modules;
- Redundant propulsion system: the vehicle is equipped not only with thrusters. The vertical translation can be performed also by means of the 2 buoyancy modules (placed one in the bow and one in the stern);
- Hovering capability: the vehicle is able to perform hovering. It has 5 DOFs (not roll) fully controllable.

MARTA is composed of several modules in Al Anticorodal housing the following components: 2 buoyancy control modules developed by AMT, 2 main vital computer ODROID-XU, 2 acoustic modems, 1 depth sensor by SensorTechnics, 1 IMU Xsens MTi-G-700 GPS, 1 DVL NavQuest 600 Micro, 1 Radio modem, 6 LiPo batteries by MaxAmps, magnetic activation...
switch, motor drivers developed by NESNE and the acoustic and optical payload devices already described in Section 2. All these components have been, or are going to be, integrated into the prototype mechanically, electrically and in low-level software. The basic functionality of many components and the waterproofness of the first prototype module (at 12 bar, i.e. at a depth of 120m) have been already verified.

4. PRELIMINARY EXPERIMENTAL DATA FROM THE FIELD

The complete version of MARTA is planned for September 2014; at the moment, the experimental activities are thus oriented on single sensors or other devices. Particularly, all the AUV acoustic sensors are ready to be tested; therefore, it has been decided to perform some preliminary data acquisition campaigns independently by the use of MARTA AUV.

The first occasion for the acoustic testing activity is the necessity of a bathymetric campaign at the Lake Roffia, San Miniato (Pisa – Italy). Lake Roffia is a small artificial basin (2x0.2km) working as an Arno river expansion; it is also used for rowing and canoeing training and competitions. The bathymetry of the lake is a strong necessity of the local community; the mapping of the obstacles, above and below the lake surface, became a fundamental safety need after the Arno river flood of February 2014 that brought a lot of objects, mainly tree trunks, in the Roffia basin. MARTA AUV will have on board different devices for navigation and acoustic localization&inspection, such as:

- Teledyne BlueView M900 (MBES);
- MTi-G-700 GPS (GPS and Inertial Measurement Unit);
- NavQuest 600Micro (Doppler Velocity Log).

These devices were tested together through their use for the bathymetry and the obstacle mapping purposes; this system has been fixed to a wooden structure in a relative pose similar to the one it will occupy on board MARTA. In this occasion, a preliminary version of the software, based on the Robot Operating System (ROS) middleware, necessary for navigation and acoustic inspection, has been also tested. The wooden structure was mounted on a small catamaran able to navigate the lake surface (Fig. 3). Fig. 4 represents the path planned to cover the whole lake surface.

![Fig. 3 – Part of the navigation sensors and acoustic payload tested during the bathymetric campaign at Lake Roffia (Pisa - Italy)](image)
The full campaign will be performed in coming weeks; actually, some preliminary tests have been performed in order to evaluate whether the selected sensors are suitable or not for the purpose. One of the images acquired through the MBES is shown below, in Fig. 5: in the upper right corner, the echo due to a tree trunk in front of the catamaran is easily identifiable.

The preliminary obtained results are encouraging both for the completion of the Lake Roffia bathymetry and obstacle mapping. They are very encouraging too considering the use of the sensors on MARTA AUV such as the navigation and acoustic inspection system and for object recognition and frontal obstacle avoidance.

5. CONCLUSIONS AND FURTHER DEVELOPMENTS

The paper describes MARTA AUV, acronym for MArine Robotic Tool for Archaeology: it is a small-sized Autonomous Underwater Vehicle (AUV) developed in the framework of the ARROWS project. The paper summarizes the main characteristics of the vehicle and preliminary experimental data from the field, concerning its acoustic payload, are given.

MARTA AUV will be ready soon and the sea trails are scheduled for September 2014.
6. ACKNOWLEDGEMENTS

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HIGH-RESOLUTION MULTIBEAM BATHYMETRY APPLIED TO UNDERWATER ARCHAEOLOGICAL RESEARCH: A CASE STUDY FROM THE LAGOON OF VENICE

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Abstract: The Venice Lagoon that surrounds the historical city of Venice has been the scene of human settlements since Roman times and innumerable archaeological remains have been discovered all over its area. The shallowness of the lagoon has for a long time prevented the use of underwater acoustics that, in general, can help to extensively and efficiently explore the bottom and sub-bottom for new archaeological discoveries. However, the recent technological development of the multibeam systems enables them to achieve very high performances also in very shallow waters. In this study, we show the first results of a six month long multibeam survey that has been carried out using a Kongsberg EM2040 DC multibeam system in the Venice Lagoon. All the channels of the lagoon and the Canal Grande in the city of Venice were surveyed. The Digital Terrain Maps (DTM) obtained from the multibeam data reach a resolution up to 5 cm giving a new insight in the underwater archaeological research in the area.

Keywords: High resolution multibeam survey, underwater archaeology, Venice lagoon

1. INTRODUCTION

In spite of the importance of the historical city of Venice, the archaeological research in the Venice lagoon (Fig1. a) and in the city has a relatively late start: the first excavation that had made use of modern methods of recovery and documentation was carried out by an Italian-Polish group on the island of Torcello in the northern part of the lagoon at the beginning of the 1960s [1] who made use of a stratigraphic approach. From the beginning of the 1970s to the 1990s the lagoon was the location of hundreds of findings in the wetlands and some of the islands of the lagoon ([3-6] and references therein) using underwater archaeology with remains dating back up to the VI century BC until the modern time. These remains were found studying archive documents, interviewing the fishermen and by an extensive visual exploration of large areas of the lagoon, supported by the use of small cores (3.5 m deep and with 2.8 cm of diameter) and by the measure of punctual electrical resistivity, ([5-6]). In the early 1990s a new phase of stratigraphic and underwater archaeology started in the Venice lagoon and in the urban area of Venice (see [7-10] and references therein). The shallowness of the lagoon has for long time prevented the use of underwater acoustics that, in general, can help to extensively and efficiently explore the bottom and sub-bottom for new archaeological discoveries and for archaeological site preservation (e.g.[11-15]).
Fig. 1a) The Venice Lagoon and the surveyed area; 1b) The bathymetry of the Canal Grande and the Bacino S. Marco; 1c) the Canale delle Scosse immediately to the North of the Malamocco inlet; 1d) the Lido inlet (grid resolution 0.5 m, vertical exaggeration 5 x)

The first acoustic survey in the lagoon of Venice was carried out in the second half of 1990s by [16]. Their surveys and those who followed ([17-21]) aimed to identify and map former
depositional environment as an integral part of work on the origin of Venice. These surveys interested mainly the mudflat areas, while small attention has been given to the tidal channel network of the lagoon. In 2013 within the National Flagship Project RITMARE, an extensive high resolution (up to 5 cm) bathymetric survey was carried out and all the channel of the lagoon were explored covering a total area of 50 km². The area is depicted in fig.1a. In fig.1b),c) and d), we show three areas of interest for this paper: in b) the area of the Canal Grande, in c) the area of the Canal de Scoasse close to the Malamocco inlet, d) the area of the Lido Inlet.

2. DATA ACQUISITION AND PROCESSING

The multibeam data were acquired with a Kongsberg EM2040 Dual-Compact MBES during a survey carried out in April 2013. The MBES was pole mounted on a 10 m long boat with 1.5 m draft. The double-head MBES has 800 beams (400 per swath) and a frequency that can range from 200 to 400 khz. During the survey the frequency was set to 360 kHz and the data were acquired in equidistance mode, ensuring more than 30 % of overlap between different survey lines. In order to provide high accuracy attitude corrections, the Seapath 300 positioning system was used provided by a Fugro DGPS correction. The sound velocity was measured continuously close to the transducers with a Valeport miniSVS sensor. Moreover sound velocity profiles over the water column were taken regularly with a AML Smart-X sound velocity profiler. The tide corrections in all the areas were obtained thanks to the hydrodynamic model SHYFEM ([22]) giving the values of water level in 93 locations of the lagoon. The model computes the sea level at each location (station) using the wind data and the sea level data from all the tidal stations in the lagoon and at the inlets as forcing factors or as data to assimilate. The error of the model in the sea level simulation at a certain station is of about 2 cm. All the corrections are referred to the local datum Punta Salute 1897. In order to obtain the DTM file of the bathymetry we used the CARIS Hips & Sips accounting for sound velocity variations, tides and basic quality control.

3. THE ERROR IN TARGET DETECTION

The accuracy of the target detection using MBES is a very important issue also in underwater archaeology. In modern systems different environmental conditions (currents, waves) should not influence positioning of underwater objects but sometimes their acoustic characteristic may disturb a target location record. In order to establish the accuracy of our instrumental setup, we chose a mooring post as a target for positioning precision measurements. The target is located in the Lido inlet (Fig. 1d) for anchoring a large navigation buoy at a depth of 13 m. It is a square piece of concrete or cast-iron (2.5m x 2.5m), protruding about 0.5m above the sandy bottom (Fig. 2). In the centre of the target there is a connector for a chain protruding around 1m above the sinker and 0.8m wide. Although we do not have pictures of this structure, sizes and heights were taken from the high resolution bathymetry of this area. Ten survey lines were made on the same day with different tides and currents. Target detection was done manually from bottom detection points of each survey line containing buoy area. The standard deviation of the centre of the target is 18.6 cm for easting and 14.4 for northing (UTM 33N zone), range of values is respectively 65 cm and 61 cm. All the detection points lie inside a centre area of the mooring post (Fig.2). Survey lines come from port and starboard sides and were collected with different boat speed (1.6-3.1 m/s) different heading and current direction. No correlations were found between these parameters and the target positions. The most important factor for the precision of target fixing is the manual process of identification of object centre. The manual procedure is probably the reason why the points are not always in the same place. The chain connecting the sinker with the buoy is
detected differently in each survey line as a narrow cloud of points up to 2m high, changing the centre of the object.

Fig. 2: A) Survey lines for the target detection (in yellow); B) Mooring block with middle the anchoring chain used for the target detection: the white points are for manual detection, the yellow points show the range and orange is the mean central position.

4. RESULTS AND DISCUSSION

The survey allowed the identification of numerous targets of potential archaeological interest, some of them are already well known like for example the wreck in Fig 3. This is a wreck of the 14th century in the Canal Grande. Grid resolution 5 cm, vertical exaggeration 5x. The arrow indicates the position of the wreck. Cables crossing the Canal are sibile. In the bottom right part is the scour created by the vaporettos (waterbuses).
the 14th century discovered by chance in the Canal Grande in 1998 by a diver putting power cables at the canal bottom and already known by the Soprintendenza per i Beni Archeologici del Veneto (Venetian Authority for the Archaeological Heritage). The shape of the wooden wreck is not well defined because it is not fully conserved. In Fig. 3 also the cables that were put down are visible and so is a deep scour created by the vaparetto (waterbus) engines at the vaparetto stops.

Following the scheme of [12], we started compiling the information and description for each anomaly into an attribute table, which can be queried within a GIS platform. Field attributes include: coordinates; classification of the anomaly (possible boat, object, etc); a description of the anomaly; dimensions of the anomaly; water depth at the anomaly’s position; maximum height of the anomaly above the lagoon bottom; description of the seabed around the anomaly; inferred energy conditions derived from features (such as bedforms, scour) observed on the lagoon bottom and some additional notes. This table will be provided to the Soprintendenza per i Beni Archeologici del Veneto (Venetian Authority for the Archaeological Heritage) for further investigation.

In Fig. 4 a second wreck is shown. In this case the water depth at the anomaly’s position is of about 7 m. It is 13.5 m long and the stern sticks out by 1.54 m (maximum height of the anomaly above the lagoon bottom) of the sediment, while the bow is about 1 m high over the seafloor. It seems to be well conserved and its shape suggests a likely modern origin although the age of the wreck is unknown and so is the cause of its sinking. In Fig. 4 also a dune field is clearly visible, with a varying wavelength of 4-7 m and height about 20-30 cm, testifying for a high energetic environment.

Fig. 4 Wreck in the Canale delle Scoasse (Malamocco), Grid resolution 20 cm, vertical exaggeration:7 x.

5. CONCLUSIONS AND FUTURE WORK

An extensive multibeam survey was carried out during 2013 in the lagoon of Venice. A field experiment was performed to assess the target detection ability of the instrumental setup (multibeam and positioning system). This experiment showed that the instruments were able to detect a target of 2.6 m with an error of 0.23 m. Out of the large high resolution bathymetric dataset, we selected two examples of targets: a wreck from the 14th century and another wreck, whose origin and how it sank are still unknown. The high resolution of the
Kongsberg EM2040DC MBES combined with 3-D visualization techniques provides an unprecedented level of detail of the targets in the Venice lagoon channels including the ability to recognize the state of preservation of the wrecks, and the impact of the wrecks on the surrounding seafloor. Visualization of these data on the Caris Hips and Sips and ArcGIS allows us to share the exploration of these artifacts with both experts and the general public. In this regards, the classification of each artifact in the bathymetric data is ongoing. These information will be then provided to Soprintendenza per i Beni Archeologici del Veneto for further archaeological investigation. This work will: a) allow the possible discovery of new archaeological sites; b) help the experts to evaluate the state of preservation of the submerged remains.

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REFERENCES


Session 14

Innovative Approaches for Characterizing Ocean Bottom Properties

Organizers: Martin Siderius, Sergio Jesus, Peter Nielsen, Jean-Pierre Hermand and Ross Chapman
EXPERIMENTAL STUDIES ON PASSIVE BOTTOM LOSS ESTIMATION FROM A COMPACT ARRAY MOUNTED ON AN AUTONOMOUS UNDERWATER VEHICLE

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\textbf{Abstract:} The seabed bottom loss (BL) is an important quantity for predicting transmission loss in the ocean. A recent passive technique for BL estimation as a function of frequency and grazing angle exploits marine ambient noise as an acoustic source. Conventional beamforming of the noise field by a vertical line array of hydrophones is a fundamental step in this technique, and the beamformer resolution in grazing angle affects the quality of the estimated BL. The technique has so far been applied successfully by using moored or drifting arrays with a length of several meters. However, nowadays a much simpler and cost-effective bottom-survey system can be conceived: Compact, sub-meter-length arrays and a low-power-consumption data-acquisition payload can be mounted on autonomous underwater vehicles (AUV), which could then map seabed reflection properties by sampling the ambient noise field as they move under the surface. The GLASS 12 and GLASS 13 experiments represent the first attempt to investigate the performance of such compact arrays for BL estimation. In this study, data collected by the AUV-mounted, 5-element, 0.4m long array are processed to compute the BL, which is then used with an inversion method to estimate the geoacoustic properties of the bottom. Rather than accurate determination of the seabed parameters, the study aims at assessing the sensitivity of this array to different bottom types, as well as illustrating the specific challenges posed by the system. Results from the two sites will be presented along with those using alternative methods (e.g., core data) for comparison.

\textbf{Keywords:} bottom reflection loss, array processing, passive sonar, autonomous underwater vehicle
1. INTRODUCTION

An accurate model of sound propagation in the local environment is required by sonar systems for effective operation, and any underwater propagation model must include information about the acoustic properties of the two ocean-waveguide boundaries: the sea surface and the seafloor. For models based on ray tracing, this can be in the form of the power reflection loss as a function of grazing angle and frequency. The research described in this document focuses on estimating the reflection loss from the bottom (hereafter also referred to as “bottom loss” — BL) by array processing of marine ambient noise, with a special focus on investigating and expanding the potential and capabilities of short arrays for this application.

When the bottom is layered (typically, one or more sediment layers overlaying a rock basement), the BL depends both on the grazing angle and on the frequency of the incident wave. Layer thicknesses and physical properties can vary dramatically within a few hundred meters, and so does the BL. The bottom properties are unfortunately very difficult to measure directly in situ, and they are typically obtained either from existing environmental databases, or by geoacoustic inversion of measured acoustic data. Only the latter process is capable of providing adequate resolution in space for accurate propagation modelling. Perhaps the most widely employed methodology for geoacoustic inversion has so far been deploying acoustic sources (such as sound projectors or explosive charges, but in some cases exploiting sources of opportunity, such as ship noise) and hydrophone arrays, measuring the acoustic field, and employing model-based matched-field processing to determine the seabed properties by minimizing the mismatch between model predictions and measurement.

Recently marine ambient noise (mainly originating from breaking waves, wind and rain at the surface) has received increased interest as an acoustic source for passive surveying of the sea bottom. Two passive sea-bottom-survey systems are the passive fathometer [1] and Harrison and Simons’ technique for bottom-loss estimation [2], on which this research work is based. One of the main advantages of this technique is that it produces BL directly, as a function of frequency and grazing angle, without requiring data inversion schemes. This is done by beamforming ambient-noise data collected by a vertical line array of hydrophones. All other parameters being equal, the beamformer angular resolution improves when the array length (and number of elements) increases. Since the BL needs to be estimated as a function of grazing angle, limitations in the angular resolution of the array result in limitations on the quality of the estimated BL. Harrison and Simons’ technique has so far been applied to data collected by moored or drifting arrays of lengths of the order of several to tens of meters. However, the recent interest in Autonomous Underwater Vehicles (AUVs) as operational platforms has quickly changed the scenario for this technique. Combining ambient-noise-based bottom survey with the versatility and simplicity of AUVs would produce an attractive, cost-effective bottom-survey system capable of covering extended areas without requiring controlling surface vessels, complex equipment or human interaction during the mission.

The possibility of AUV deployment has made rigid, short arrays increasingly attractive, but their poor angular resolution represents a significant drawback for the purpose of bottom-loss estimation. The GLider Acoustics Sensing of Sediments (GLASS) experiment represents the first attempt at investigating on-the-field AUV-based bottom survey through ambient-noise processing. The NATO-STO Centre for Maritime Research and Experimentation (CMRE, La Spezia, Italy) has developed a rigid, compact array specifically for this purpose,
which has been mounted on the nose of an AUV and deployed at sea in two different experiments in 2012 and 2013.

The work described in this paper presents the results of BL estimation by processing of the GLASS data, and employs the results as input to a model based inversion-algorithm to estimate the physical properties of the two bottom types. The aim of this study is a first assessment of the sensitivity of this array to different bottom types. Furthermore, the study offers an opportunity for illustrating the specific challenges posed by the system, and for investigating techniques that can help overcome such challenges.

2. METHOD

The goal of the GLASS project is to apply Harrison and Simons’ technique, so far implemented only using moored or drifting arrays, to data collected by a compact array mounted on an autonomous vehicle. For this purpose, CMRE developed a nose-mounted combined vertical and tetrahedral array (see Fig.1), comprising a total of 8 elements, 5 of which are on a straight (vertical) line with 0.1m spacing. The central element of the line array is also one of the vertices of the tetrahedral array, which includes the remaining 3 elements; the spacing between any two sensors in the tetrahedral array is still 0.1m.

The AUV with mounted data-acquisition package was deployed for the first time in the Summer of 2012 (GLASS’12) off the Versilian Coast, Mediterranean Sea (Italy), and a second time in the Summer of 2013 (GLASS’13) off the coast of Panama City, Florida, (USA). The scientific goal was to provide experimental validation of the performance of an AUV-mounted short array in ambient-noise-based bottom characterization.

Before showing the results of the study, it seems useful to introduce the challenges that such a short array presents for this particular application. The plots in Fig.2 show the BL predicted by a theoretical model (Jensen et al. [3]) and computed by beamforming simulated data obtained using OASN [4], a software package implementing a numerical solution to the full wave equation for range-independent, stratified media. For this reference case, the bottom is a halfspace and the array has the same inter-element spacing as the GLASS array, with varying number of elements.

One of the goals of this experiment was to investigate whether, despite the deterioration due to the reduced array length visible in Fig.2, the bottom-loss estimate produced by such and array can still be used for inferring useful information on the bottom. For this purpose, the system was deployed at sites with different bottom characteristics. Based on previous experimental activities and core data acquired during the GLASS’12 experiment, the seabed properties in part of the GLASS’12 experimental area can be characterized as clay or clay-silt. The GLASS’13 site has been visited several times in the past, and independent measurements of the bottom properties were obtained during the Target and Reverberation Experiment 2013 (TREX13), conducted in the same area. Results from these experiments indicate that the seabed is composed mainly of sand with shell debris and clay/silt inclusions. The plots in Fig.3 show examples of the BL obtained from GLASS data in the two campaigns. The marked differences in the two plots show that the measured loss is clearly affected by the different bottom types.

The second part of this study aimed at obtaining more quantitative results, by investigating the feasibility of inverting the data for geoaoustic parameters of the bottom. The estimate obtained from data collected during the two GLASS campaigns were used as input to a model based inversion algorithm in order to estimate the physical properties of the two bottom types. In this analysis, the seabed is described as an infinite halfspace with only three unknown parameters, namely density, attenuation and sound speed. The inversion is based on
a simple algorithm, which does not aim at competing with state-of-the art systems: The aim of this study, rather than accurate determination of the seabed parameters, is a first assessment of the sensitivity of this array to different bottom types.

![Fig.1: The GLASS array mounted on the nose of a compact AUV — Side and front view.](image)

The main components of the search algorithm are the cost function and the search space. The algorithm compares the BL estimated from data at all grazing angles and in the 1-7.5kHz frequency range to the entire corpus constituting the search space and produces as solution the bottom configuration that results in the lowest cost-function value. The cost function is the RMS of the pixel-by-pixel difference between the data-estimated BL and the model-derived BL for a given combination of the bottom parameters. The search space included bottom configurations given by all possible combinations of the discretized values of the three search parameters: 8 values for the attenuation covering the range from 0.2 to 1.6 dB/λ (where λ is the signal wavelength); 15 values for the sound speed in the range from 1440 to 2000 m/s, and 41 density values in the range from 1000 to 3000 kg/m³, for a total of 4920 configurations constituting the search space.

The corpus constituting the search space was obtained by running for each configuration the following procedure:

1. Jensen et al.’s model [3] was implemented and run to compute the plane-wave power reflection coefficient for each combination of the bottom parameters.
2. To compute the spatial coherence function between each pair of hydrophones, Harrison’s model [5] was implemented. This receives as input the reflection coefficient generated in step (1), together with the water-column physical parameters and positions of the array hydrophones.
3. The coherence function resulting from Harrison’s model is used to build the cross spectral-density (CSD) matrix, which is then beamformed by a conventional beamformer (CBF) and applying a Taylor taper to the array.
4. The result is used to estimate the BL with Harrison and Simons’ technique [2].

Harrison’s model was preferred to OASN in this study for performance reasons, as its implementation has much lower running times. Excellent agreement between the implemented models and OASN has been verified by comparing the results over a few test cases (which included layered bottoms).
Fig. 2: Halfspace bottom: Bottom loss as a function of grazing angle and frequency as predicted by Jensen’s model (top left) and estimated from OASN data using an array with 40 (top right), 10 (bottom left) and 5 elements (bottom right).

Fig. 3: Bottom loss estimated from GLASS data. The data samples are from the 2012 (left) and 2013 (right) campaigns.
3. RESULTS

Two examples of the results obtained by the exhaustive search algorithm for GLASS’12 and GLASS’13 data are shown in Table 1 and Fig.4. The sensitivity of the search to the three parameters has been examined in a number of cases, showing consistently that the sound speed is the most sensitive parameter, i.e. the one that induces the largest variations of the cost function for a given configuration, followed by the density and the attenuation.

![Fig.4](image)

**Fig.4** Bottom loss predicted by the theoretical model for the two bottom configurations chosen by the inversion algorithm for the two corresponding data samples in Fig.3.

<table>
<thead>
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<th>Data file</th>
<th>Optimal $\alpha_p$ (dB/$\lambda$)</th>
<th>Optimal $c$ (m/s)</th>
<th>Optimal $\rho$ (kg/m³)</th>
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<td>1560</td>
<td>250</td>
</tr>
<tr>
<td>glass13_2013_06_13_17_43_08</td>
<td>0.6</td>
<td>1600</td>
<td>1750</td>
</tr>
</tbody>
</table>

**Table 1**: Results of GLASS data inversion for two data samples.

The procedure was repeated for other data snapshots at the two sites, showing results consistently in agreement with those reported in Table 1. For GLASS’12, the sound-speed value is in agreement with the results of core measurements from the experiment, whereas the density appears too high. One hypothesis is that this could be due to the low wind-noise level at the site, a condition that has been indicated as capable of hindering the results of this BL estimate technique [6]. For GLASS’13, both the sound speed and the density value are in good agreement with estimates obtained at the site.

4. FUTURE DIRECTIONS

The plots in Fig.2 show that one way of improving the BL estimate would be to increase the array length, but this comes at the price of greater weight and drag for the AUV, as well as more complex requirements for on-board data collection and storage, and higher power consumption. A promising alternative is synthetic-extension array processing, which can...
improve the angular resolution of the array by exploiting the physical characteristics of the
noise field [7] [8]. Another possible approach is coherence-function extrapolation [9], to
which a new simplified approach is shown here. A theoretical expression for the spatial
cohereence function between two hydrophones in a vertical-line array is given by Harrison [5]:

\[
C_\omega(z) = \frac{\pi^2}{2} \sin \theta_s \cos \theta_r \int_0^{\infty} \frac{2}{1 - R_s(\theta_s)R_b(\theta_b)} e^{-2\omega c z} \left\{ e^{-i\omega z} e^{-\alpha z} + R_b(\theta_b) e^{-i\omega z} e^{-\alpha z} \right\} d\omega.
\]

(1)

In Eq. (1), \(C_\omega(z)\) is the coherence function for the hydrophone pair, assumed to be aligned
with the \(z\) axis, with the first hydrophone at \(z = 0\). Furthermore, \(\theta_r, \theta_s,\) and \(\theta_b\) are the ray
angles at the receiver, the surface, and the bottom; \(s_c\) and \(s_p\) are the complete and partial
ray-path lengths, whose dependence on \(\theta_r\) is determined by the sound-speed profile in the
water column; \(\omega\) is the angular frequency; \(c\) is the sound speed at the receiver in the medium,
and \(R_s\) and \(R_b\) are the bottom and surface power reflection coefficients. For the sake of
simplicity, the dependence of the reflection coefficients on frequency is not indicated
explicitly. Note that \(a\) is the power attenuation per unit length.

For relatively simple bottoms, neglecting the effect on \(C_\omega(z)\) of the frequency dependence
of \(R_b\) can be an acceptable assumption in order to extrapolate the coherence function. When
this assumption is made, the dependence of the coherence function on the sensor spacing \(z\)
lies primarily in the two exponentials, where \(z\) always appears multiplied by the angular
frequency \(\omega\). This means that multiplying \(z\) by an integer factor (as is done to obtain the coherence function between different hydrophone pairs in the array) has the same effect as
leaving \(z\) unaltered and multiplying \(\omega\) by the same factor. The preceding statement can be
exploited to extrapolate the coherence function measured by an \(M\)-element array: The
maximum spacing for which the coherence function can be measured from data is
\(z = (M - 1)d\), and the (extrapolated) value of the function at \(z = nd (n \geq M)\) can be obtained
by assuming:

\[
C_\omega(nd) \approx C_{\omega/(M-1)}[(M - 1)d].
\]

(2)

The results of this technique on data from an OASN simulation with a single sediment
layer over a rock halfspace are shown in Fig.5 and show interesting potential for this study.

5. CONCLUSIONS

This paper presents results of BL estimation by processing of marine ambient noise
collected by an AUV-mounted, 0.4m hydrophone array. The results from two different sites
indicate that such a short array is sensitive to different bottom types and produces results in
satisfactory agreement with available ground truth data. However, it would be desirable to
improve the grazing-angle resolution of the results. This can be done by increasing the array
length or, when this is not possible, by use of processing techniques such as synthetic-
extension array processing or coherence-function extrapolation. A simple approach to the
latter is introduced, showing very encouraging results in simulation.
Fig. 5 Single sediment layer: CBF-estimated bottom loss for 10-element array extended by coherence-function extrapolation to 20 elements (left), and for a 20-element array extended to 40 elements (center). The latter result is very close to that of the full 40-element array (right).

6. ACKNOWLEDGEMENTS

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Abstract: The Centre for Maritime Research and Experimentation conducted the Glider Acoustic Sensing of Sediments experiment in the Mediterranean Sea in the summer of 2012. Among the objectives of the sea trial was to employ a new autonomous underwater vehicle to collect acoustic data to invert for geoacoustic properties of the seafloor sediments. Acoustic data were collected with the vehicle bottom moored in 18 meters of water while the R/V Alliance made a close pass at five meters per second. Data from a single hydrophone channel were inverted for the compression wave speed and attenuation in the seafloor sediments. The inversion process operated on broadband received levels recorded during the pass-by of the Alliance and the ambient noise spectrum while the Alliance loitered four kilometres distant. Parameter estimates were generated using an evolutionary algorithm to minimize an objective function that was developed for this study and a forward model based on acoustic normal modes. The model vectors generated during the optimization were post-processed using a maximum-likelihood approach to arrive at the parameter estimates and uncertainties. Good agreement was found between the compression wave speed estimated by the inversion and measurements performed on two geophysical cores taken in close proximity to the bottom moored vehicle. The broadband source spectrum of the Alliance was also estimated across more than six octaves of frequency.

Keywords: geoacoustic inversion, differential evolution, maximum likelihood
1. INTRODUCTION

Estimation of seabed sediment properties by inversion of acoustic field data has been the subject of considerable attention. A significant portion of that work resulted in different methods for the inversion of the noise radiated by surface ships for various properties of the seabed. [1] An approach often used has been to exploit the time-frequency characteristics of data collected by an array of hydrophones oriented vertically or horizontally with respect to the seabed. [2] Inversions performed with data collected at a single point [3] and over a broad spectrum of frequencies [4] are less common.

The objective of the inverse problem presented here was to simultaneously estimate properties of the acoustic source and the environment, using data collected by a single hydrophone. The acoustic source spectrum of a surface ship passing near a bottom moored autonomous underwater vehicle (AUV) was estimated in contiguous 1/3 octave bands spanning more than six octaves of frequency. In addition, the compression wave speed and attenuation in marine sediments near the AUV were estimated for a total of 21 unknown parameters. While the compression wave speed estimates compared well with measurements performed on geophysical cores, the process did not converge to an estimate for sediment attenuation.

2. EXPERIMENT

Data for this study were collected during the Glider Acoustic Sensing of Sediments 2012 (GLASS’12) experiment that was conducted in the Mediterranean Sea at the location shown in Fig. 1. The data acquisition system included the acoustic array, signal conditioning, digital recording and data storage. The system collected eight channels of digital data at 100 kHz with 24-bit precision from an array (see Fig. 2a) mounted to the

Figure 1: GLASS’12 experiment site a) approximately 10 km off the Italian coast near Viareggio, and b) during a close pass by the R/V Alliance. Tick marks on ship’s track provided at two minute intervals.
nose of the vehicle. The geometry of the array elements formed two distinct acoustic apertures. The first was a five element vertical aperture with 10 cm spacing. The second was a four element tetrahedral aperture that shared one element with the vertical.

Ship radiated noise data were collected during a close pass of the Alliance (see Fig. 1b) with the vehicle bottom moored in 18.5 meters of water using a purpose built test stand (see Fig. 2b) that maintained the vehicle in a horizontal orientation with the longitudinal axis located about 1.5 meters above the seafloor. The water column was characterized by a typical downward refracting summer sound speed profile as illustrated in Fig. 3a. The data set began at the closest point of approach (CPA) and extended for less than two minutes as the Alliance proceeded south at about 5 m/s. Figure 3b illustrates received level data in the 250 Hz band during this event. (Also shown is the output of the forward model using parameter estimates generated by the inversion as will be discussed in Section 4.) Acoustic data representing the ambient and self-noise spectrum observed when the Alliance was loitering at a distance of four kilometers were also used (see Fig. 3c). The elevated noise levels in the 200 and 400 Hz bands were due to the power supply used for the hydrophone array.

Received acoustic band levels were estimated from data records that were five seconds in length with adjacent records overlapping by 50%. Each five second observation was decomposed into an ensemble of short buffers (e.g., 0.328 seconds) from which acoustic power spectra were computed. Due to a moderately active population of shrimp, the median value of each 1/3 octave band level was used as the best estimate, the median being a more robust estimator than the mean when contending with noise from individual snapping shrimp. The spread of each observation was represented by the interquartile range for the same reason. Thus, the inversion data set included the location of the AUV, the Alliance track (Fig. 1b), the local sound speed profile (Fig. 3a), the noise radiated by Alliance as range from the AUV increased from 75 to 550 meters (Fig. 3b), and the ambient noise observed in the absence of Alliance (Fig. 3c).
3. INVERSION METHOD

3.1. Forward Model

The inverse method developed for this study treated the acoustic source, the Alliance, as an unknown subject of the parameter estimation problem, in addition to certain properties of the seabed sediments. The acoustic source was modeled as compact and located at a depth of three meters, corresponding to the propeller depth. The forward model for the inversion predicted the received acoustic pressure in 1/3 octave band levels. The sound speed profile, the source-receiver positions and sediment density were known quantities, the latter from geophysical core data. Thus, the forward model for the acoustic pressure amplitude \( P_o \) observed at frequency \( f \) was represented by the Alliance source spectrum \( s_P \) convolved with the magnitude of the acoustic transfer function \( H \) between the source and receiver separated by the horizontal distance \( r \), together with the combined ambient and self-noise pressure amplitude \( P_a \)

\[
P_o^2(f, r) = s_P^2(f) H^2(f, r, c_b, a_b) + P_a^2(f),
\]

where the compression wave speed and attenuation coefficient in the seafloor sediments were \( c_b \) and \( a_b \), respectively. The seafloor was modeled as an acoustic half-space. Dependence of the acoustic transfer function \( H \) on the sediment density and sound speed profile in the water is implicitly assumed in Eq. (1). Frequency dependent, acoustic transfer functions were computed using the normal mode model KrakenC. [5] The magnitudes of the acoustic transfer functions were estimated as the average value for ten discrete frequencies linearly distributed over each 1/3 octave band.
3.2. Nonlinear Optimization

An objective function was defined to guide the directed search for the set of model parameters that best explained the observed data. The data on which the objective function operated were the acoustic pressure amplitudes in contiguous 1/3 octave bands ranging from 125 Hz to 8.0 kHz observed at 42 source-receiver separations between 75 and 550 meters for a total of $N = 798$ observations. The error associated with the $n^{th}$ frequency-distance pair in the $[N \times 1]$ error vector $e$ was

$$ e(n,m) = \ln \left( \frac{P_m(n,m)}{P(n)} \right), $$

where $P_m$ was the acoustic pressure predicted by the forward model for parameter set $m$ (e.g., source spectrum, sediment compression wave speed and attenuation). An objective function based on this error vector was defined as

$$ \phi(m) = \left( (w \cdot e(m))^T (w \cdot e(m)) \right)^{1/2}, $$

with $T$ the transpose operator. The weight vector $w$ provided observations with lower interquartile ranges with greater influence in the objective function value such that

$$ w = \frac{N}{\sum_{n=1}^{N} P_{on} q_n} \left[ \frac{P_{o1}}{q_1}, \frac{P_{o2}}{q_2}, \ldots, \frac{P_{oN}}{q_N} \right]^T, $$

where $q_n$ was the interquartile range of the $n^{th}$ observation.

A global search strategy based on the differential evolution [6] algorithm was used to minimize Eq. (3). The optimization was initialized with a population randomly distributed throughout a bounded parameter search space $M$. Source level bounds were centered on the spectrum observed at CPA plus the transmission loss for a spherically divergent wave field. The search bounds spanned $\pm 20$ dB about this reference value, plus an additional factor for the interquartile range of the data. Parameter bounds for sound speed in the bottom were 1000 to 2000 m/s. The attenuation coefficient $a$ was bounded at 0 to 1, where the frequency dependent attenuation was

$$ \alpha(f) = \frac{a f}{10^4} \text{dB/m}, $$

with $f$ the frequency in Hz.
3.3. Parameter Values and Uncertainty Estimates

Parameter values and their uncertainties were expressed using *a posteriori* probability distributions derived from a maximum likelihood approach [9] where the *a priori* $U$ and *a posteriori* $G$ probability distributions were related through a likelihood function $L$ as

$$G(m) = L(m)U(m),$$

and the one-dimensional marginal *a posteriori* probability density function for the $i^{th}$ parameter $G_i(m_i)$ was the integral of the $M$ dimensional probability density with respect to all parameters $m_j$ for $j = 1,2,\ldots,M$ and $i \neq j$ to yield

$$G_i(m_i) = \int \cdots \int G(m) \, dm_1 \cdots dm_{i-1} \cdots dm_{i+1} \cdots dm_M. \tag{7}$$

Methods to estimate the integrals of Eq. (7) include importance sampling where the integrand is sampled non-uniformly to concentrate the computational effort to regions that contribute most to the integral. The differential evolution algorithm implements importance sampling using a generating distribution to select the trial model vectors. While the distribution is unknown and evolves over the course of the optimization, a large number of candidate solutions are generated from the total model parameter space $M$ during the search process. Thus, the model vectors $m$ and objective function values $\phi$ computed during the search can be used to approximate the integrals of Eq. (7). [10]

An estimate for the *a posteriori* probability of the $k^{th}$ model vector based on the $N_p$ model vectors in the population at the conclusion of the optimization process is

$$\hat{G}(m_k) = \frac{L(m_k)U(m_k)}{\sum_{j=1}^{N_p} L(m_j)U(m_j)}, \tag{8}$$

with the *a priori* probabilities $U$ uniformly distributed between the bounds defined for each element in the model parameter vector $m$. The marginal probability distribution for obtaining the particular value $K$ for the $i^{th}$ parameter $m_i$ in the model vector is

$$\hat{G}_i(K) = \sum_{k=1}^{N_p} \hat{G}(m_k) \delta(m_{k,i} - K). \tag{9}$$

Since, the likelihood function is usually related to the objective function $\phi(m)$ through an exponential, an empirical estimate is
\[ L(m) = \exp\left(-\frac{\phi(m) - \phi(m_o)}{T}\right), \] (10)

where \( m_o \) is the parameter vector for the optimum value of the objective function and \( T \) is a constant that is particular to each optimization. A common value for \( T \) is the average of the 50 best objective functions obtained during the optimization minus the best value of the objective function. [9]

4. RESULTS

Inversion results for the source spectrum and sediment compression wave speed are provided as Fig. 4a and 4b, respectively. Parameter estimates are presented as the mean plus and minus one standard deviation for the \emph{a posteriori} probability distributions. Figure 3b compares the received levels observed in the 250 Hz band to those calculated by the forward model using the set of model parameter estimates.

![Figure 4: GLASS’12 inversion results a) R/V Alliance source spectrum estimate, and b) sediment compression wave speed estimate. Error bars indicate plus and minus one standard deviation for the \emph{a posteriori} probability distribution estimates.](image)

The source spectrum estimates include the search bounds used during the optimization and a reference spectrum for a spherically divergent acoustic field between the Alliance and the bottom moored AUV while at CPA. The relatively large uncertainty in the source level estimated for the 200 Hz band (\( \sigma = 8.5 \text{ dB} \)) was due to the high level of electronic noise in the data recording system (see Fig. 3c). The standard deviations of the estimated source level distributions were otherwise on the order of two to three decibels.

The estimated compression wave speed is illustrated in Fig. 4b. Also shown in the figure are the compression wave speeds measured in two cores taken within 500 meters of each other and the AUV deployment location. The compression wave speed estimate of
1529 m/s ($\sigma = 10$) agrees well with the mean value of 1510 m/s ($\sigma = 15$) that was measured in the cores. The inversion did not converge on an estimate for attenuation in the sediment, its influence on the value of the objective function being negligible.

5. CONCLUSION

Acoustic data collected by a hydrophone mounted to a bottom moored AUV were inverted for the broadband source spectrum of a passing ship and the compression wave speed in the seafloor sediments. An evolutionary algorithm was used in conjunction with a maximum likelihood approach to estimate the set of model parameters with good results. The process did not converge on an estimate for sediment attenuation.

6. ACKNOWLEDGEMENTS

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Session 15

Modeling Sonar Performance in Uncertain Environments

Organizers: Georgios Haralabus and Chris Strode
MITIGATION METHODS AND TECHNIQUES FOR ENHANCING SONAR OPERATIONAL CONFIDENCE

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Abstract: There are numerous challenges to effectively model sonar performance given the variability in the ocean acoustic parameters, and the uncertainties with respect to factors such as the target strength. As such, in most cases it is well recognized by the operational community that it is nearly impossible to accurately predict the performance of a sonar system for the full range of operational settings. Given this situation, numerous initiatives both technical and operational are required to harmoniously provide a level of confidence in sonar prediction under practical conditions. Some initiatives and tools that are being utilized include the NATO Multi-Static Tactical Planning Aid (MSTPA) and the Environmental Modelling Manager (EMM). The latter is embedded in a developmental version of the Canadian Maritime Acoustic Processing System (MAPS). These are presented with the caveat that some may have a greater or lesser impact on operator confidence under realistic operational conditions.

Keywords: sonar, operational, modelling
INTRODUCTION

There are numerous challenges to effectively model sonar performance given the variability in the ocean acoustic parameters, and the uncertainties with respect to factors such as the target strength. As such, in most cases it is well recognized by the operational community that it is nearly impossible to accurately predict the performance of a sonar system for the full range of operational settings.

For practical purposes, numerous initiatives both technical and operational are required to harmoniously provide a level of confidence in sonar prediction under practical conditions. In other words, the operator still requires some measurement to enable operational planning and tactical analysis, and to measure the risk associated with not being able to get underwater contacts. Some of these items are discussed next.

ENVIRONMENTAL UNCERTAINTY

The passive or active sonar equations require values for each of the terms to achieve a solution; and are generally input as estimates. Each of the parameters is expected to be a median or mean value, however, variability and uncertainty in the environmental parameters as well as the target strength or target noise levels are such that there is an expected variability between the actual observed signal excess $SE_{obs}$ the expected or modelled value $SE_{exp}$ [1];

$$SE_{obs} = SE_{exp} + e$$

(1)

The uncertainty in signal excess typically results from uncertainty in the sound speed profile and boundary conditions (sea surface and sea bed). This can result from both measurement errors/approximations and an incorrect assumption that conditions at a measured location may be applied throughout a scenario area. With scenario areas often in the order of 100 by 100km there may be significant variation in sound speed profile, bottom type, and surface wind speed from one region to another. All of these factors serve to complicate the acoustic transmission paths and change both reflection loss and scattering strength. Monte Carlo simulations were conducted in [2] with sound speed profiles, surface, and bottom conditions sampled according to estimated uncertainty in modelled parameters. The results demonstrated significant variations in signal excess which were seen to be range dependent over a scenario area.

Sensitivity analysis may be used to determine which parameters contribute most to the uncertainty in signal excess. For example in a downward refracting environment the bottom parameters – such as density, attenuation, and sound speed – will contribute most to the uncertainty. This type of analysis may be used to direct military oceanography to measure the most important part of the environment and is the focus of the Environmental Knowledge and Operation Effectiveness (EKOE) program at CMRE [3].

The difference between the modelled and observed SE can be measured by doing real-time, real world experiments and operational measurements of SE. Ideally, this would be provided within the acoustic operator’s station so that real-time measurements may be incorporated immediately into acoustic prediction models. Efforts at DRDC are underway to provide this in the Maritime Acoustic Processing System (MAPS). MAPS is based on
DRDC Atlantic’s open architecture developmental System Test Bed (STB) and has been installed as an experimental system on various Canadian Patrol Frigates (CPF). Ideally, MAPS would provide both a higher fidelity prediction of the acoustic conditions for immediate and accurate understanding of the local environment while MSTPA will provide a larger area tactical prediction tool of the evolving situation.

ENVIRONMENTAL MODELLING MANAGER (EMM)

The Environmental Modelling Manager is an environmental tool embedded within MAPS. In order to get an estimate of transmission loss, one of the key parameters in the sonar equation, an estimate of the sound speed is required. The sound speed is determined typically with the aid of a sound speed profile measurement. This includes the temperature profile, bottom depth as well as a measurement of salinity. Figure 1 shows the transmission loss estimate data interactions. Figure 2 shows the propagation path display, while Figure 3 shows the transmission loss display.

![Fig.1: Transmission Loss Estimate Data Interactions](image1)

![Fig.2: Propagation Path Display](image2)
NATO MULTI-STATIC TACTICAL PLANNING AID (MSTPA)

The NATO Multi-Static Tactical Planning Aid (MSTPA) is one of several developments from CMRE. The purpose of the aid is to develop a means whereby scientists could assess the performance of both mono-static and multi-static sonar systems. In addition, one of the goals of the aid is to provide a tool that operators might use for tactical planning purposes. To that end, a NATO international team has been formed to examine the use of the tool for operational guidance.

The MSTPA main program scenario editor contains various window panes and menus with a world map viewer. The left window pane contains the separate elements that are used in a tree-view format under the main level Project element to enter the different aspects and attributes of the scenario. These include the following:

1. Scenario Area;
2. Environment;
3. Target Location Sampler and False contact definition;
4. Tracker parameters definition;
5. Labeler- for Poss-sub and Non-sub definitions;
6. Multistatic System- this essentially comprises the ships, sonobuoys, and aircraft sources and receivers in the multistatic sensors;
7. Targets;
8. Output files and active windows to be displayed during the simulation and for post-analysis; and
9. If required, an optimiser, to be run with another version of MSTPA - the MSTPA Optimiser.

The acoustic environment within MSTPA may be quickly configured and can include simple flat bottom isovelocity or fully range dependent sound speed profile and bathymetry. Furthermore, bottom and surface parameters may be configured allowing for an assessment to be made of their impact on signal excess. Furthermore the ARTEMIS acoustic model – employed within MSTPA – may be configured to operate in one of two different modes. Firstly, fast range dependent propagation may be modelled employing an adiabatic incoherent approximation. This provides fast results but sacrifices some of the fine detail arising from coherent propagation. With a single switch the model can include the coherent propagation – increasing fidelity – at the expense of calculation time [4]. Figure 4 shows probability of detection in range and depth for a range dependent environment using the fast incoherent mode. Figure 5 shows probability of detection for identical parameters but with the coherent effects included. Given the uncertainty in sound speed profile and boundary conditions considered here it is likely that the fast incoherent
results will be sufficient for most operational range predictions. In February 2014 MSTPA was deployed at sea during the Dynamic Mongoose ASW exercise. During the exercise the MSTPA incoherent predictions agreed well with those made by the Norwegian Navy.

One of the main outputs of MSTPA is the generation of Signal Excess (SE) plots, which corresponds to $SE_{obs}$ from equation (1). An example SE plot is shown in Figure 6 against a fictitious submarine. In such a plot the environment is gridded and a target placed in each cell. Full range dependent acoustic calculations are the made to every target over the scenario area of interest. Such a plot can provide an effective graphical assessment of both extent and variability in detection performance.

During a recent NATO Proud Manta 2013 exercise, MSTPA was used aboard the research vessel ALLIANCE to generate the plot in Figure 6. When the actual submarine track shown in Figure 7 was obtained and incorporated into MSTPA, the results showed where a likely detection would occur.
As mentioned, MSTPA was deployed at sea during the Dynamic Mongoose ASW exercise in 2014. The tool was installed on a Laptop and placed in the operations room of the SNMG1 command ship. With MSTPA running in its real time mode the platform positions within the simulation could be constantly updated with actual ship positions. In this way coverage plots for all platforms within a Task Group could be determined at any time according to the actual disposition of assets.

The exercise was conducted off Norway and was a dynamic and complex acoustic environment including fresh water run-off from Fjords and rapidly changing sea state and wind speed during the course of a typical serial of around 10 hours. The effect of the environment on predicted sonar performance is shown in Figure 8 in which probability of detection is plotted for two platforms with identical hull mounted sonars. It is apparent that the southerly platform has significantly degraded performance. Further investigation
shows that this platform is in a region of lower sound speed as shown in Figure 9. The fully range dependent sound speed information was downloaded from the myOcean website which runs oceanographic models to forecast sound speed profile throughout the world. The actual platform position at the time of the sonar performance prediction corresponded to the northerly platform. The platform had at this point measured the sound speed profile and generated its own range prediction. However, it would not have captured the effect of the changing underwater environment without the use of a fully range dependent acoustic tool such as MSTPA.

**Fig.8:** Performance prediction for two identical sonars within Dynamic Mongoose exercise area

**Fig.9:** Surface sound speed
SUMMARY AND CONCLUSIONS

With increasing capabilities of both oceanographic measurement devices and propagation models – sonar performance predictions may be generated while accounting for spatial and temporal variations in the underwater environment. However, forecasts of sound speed profiles and boundary conditions are subject to a degree of statistical uncertainty which must be accounted for in any results provided to ASW operators. The tools detailed in this report may be used to model multiple instantiations of possible environments in order to demonstrate the degree of uncertainty in sonar performance predictions. Further work at CMRE and DRDC Atlantic will determine ways in which this additional information may be effectively displayed on-board assets in conjunction with range predictions.

REFERENCES

PROPAGATION OF ACOUSTIC WAVES THROUGH A SPATIALLY FLUCTUATING MEDIUM: THEORETICAL STUDY OF THE PHYSICAL PHENOMENA.

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Abstract: The authors focus on the effects of phenomena, such as linear internal waves, that are responsible for fluctuations of the depth-dependent sound speed profile and, hence, induce distortions of the resulting acoustic pressure field and degradation of the associated sonar performances. The main goal of this study is to develop a scaled experiment configuration able to provide some results representative of this kind of distortions. To do so, a theoretical study of the phenomenon has first been carried out: we obtained an expression for the standard parabolic equation applied to the Fourier transform of the moments of order 2 and 4 in 3D medium.

Various simulation programs were developed and used for the following purposes: validating or discarding some relationships given by Flatté through his classical dimensionless analysis (\(A\Phi\) plane); tracing rays through an acoustic lens featuring a plane face and a randomly rough face and propagating an acoustic wave through the same object in order to anticipate for the shape of the distorted pressure field, including diffraction effects.

We were able both theoretically and experimentally to induce acoustic scattering that mimics, at reduced scale and frequencies around 2MHz, the correlation properties and the corresponding array performance that would be observed at sea, after propagation through a linear internal wave field, or reflection on a rough sea surface.

Keywords: De-coherence, Tank Experiments, Fluctuations, Dimensionless Analysis.
1. INTRODUCTION.

Wave propagation in random media (WPRM) is a well-studied topic in the literature. From a historical point of view [1] [2], to more recent applications to various fields such as optics [3] or underwater acoustics [4] [5], it has been proven that, in order to enhance the performances of operational systems, stochastic knowledge of the fluctuating media is needed. Linear internal waves (LIW) are an example of fluctuations in the ocean medium that causes perturbations in the underwater sound propagation, such as the appearance of caustics [6].

The performance of acoustic arrays associated with this kind of phenomena is therefore mitigated [7-10]. The objective of our research is to perform measurements of sound waves fluctuations in a controlled environment (water tank), which, in our opinion, would help to understand the involved physical phenomena. In a second time, corrective signal processing techniques are sought out.

In this paper, we focus on the theoretical approach and present some simulation results compared with experimental data acquired in a tank.

2. THEORETICAL STUDY.

We first wrote the standard parabolic equation applied to the Fourier transform of the 2nd and 4th order moment of the pressure field (respectively denoted $J_2$ and $J_4$ (extension from 2D to 3D media)). These equations and the results associated (expression for the average number of eigen rays) are presented in [11].

The method we adopted for reproducing the effects of LIW like phenomena in water tanks is the propagation of an ultrasonic signal through an acoustic wax lens presenting a plane input face and a randomly rough output face. The randomly rough face of the lens is characterized by its amplitude $\delta_x$ and its vertical and horizontal correlation lengths (respectively $L_v$ and $L_H$). This random profile is represented in Fig.1.

![Randomly Rough Output Face Profile](image1)

*Fig.1: (a): Randomly Rough Output Face Profile - $\delta_x = 3mm$; $L_v = 3mm$; $L_H = 30mm$; (b) CAD view of the Lens.*
If we can anticipate quite reasonably that our experimental configuration would induce focal points, caustics and distortions of the acoustic wave fronts, a way to link our work to the theory developed in the literature is needed. Flatté defined two dimensionless parameters in order to classify the acoustic signal distortions into perturbations regimes: the diffraction parameter \( \Lambda \), qualitatively describing the nature of the acoustic distortions, and the strength parameter \( \Phi \), characterising the amplitude of these distortions. Both parameters depend upon oceanographic quantities that are measurable during at-sea experiments [4]. Unfortunately, we cannot evaluate directly these parameters with the configuration given in our experimental framework.

Flatté proposed connections between these two parameters and the average number of eigen rays \( \langle N_{eig} \rangle \), the rms phase difference between the extreme micropaths \( \Delta_{\text{RMS}} \) and the total vertical range over which the micropaths are spread \( \Delta_z \) [4]:

\[
\Lambda \Phi \approx \langle N_{eig} \rangle \quad (1), \quad \Lambda \Phi^2 \approx \Delta_{\text{RMS}} \quad (2), \quad \Lambda \Phi \approx \frac{\Delta_z}{L_V} \quad (3)
\]

Our first goal was therefore to validate these equations: we developed a simulation ray tracing program in order to assess the parameters involved in the right members of equations (6) to (8) in the case of a medium presenting Gaussian fluctuations of the sound speed. The software is based on tracing rays and takes into account the local sound speed fluctuations. The ray tracing system from [12] is solved using a 5th order Runge-Kutta technique [13].

![Ray Trace - Medium Presenting Fluctuations of the Sound Speed](image)

**Fig.2:** (a): Ray Trace - Medium Presenting Fluctuations of the Sound Speed - \( f=1kHz \) – \( z_S=500m - 404m < z_R < 596m \) – \( 1 \text{ km} < d_i < 15 \text{ km} \) – \( L_V=50m \) – \( L_H=500m \) – \( \delta_c=1m/s \).

(b) to (d): Equation (6) to (8) Validation.

Fig.2 shows a relatively good agreement between our simulations and Flatté's identities corresponding to equations (6) to (8). All three relations can be validated and used in the following work.
3. SIMULATION RESULTS.
3.1. RAY TRACING AND Λ-Φ PLANE.

With the ray tracing program, it was possible to predict the average number of eigenrays $\langle N_{eig} \rangle$ and the rms phase difference between extreme micropaths $\Delta_{\phiRMS}$ when a beam of rays propagates through the wax lens defined earlier. These predictions were conducted for several positions of the sensor used as a receiver. The distance between the source and the object is also a tunable parameter. Therefore, we were able to evaluate $\langle N_{eig} \rangle$ and $\Delta_{\phiRMS}$ for virtual arrays (either vertical or horizontal) at various distances of the distorting object.

![Ray Trace – Vertical Direction – Wax Lens.](image)

![ΛΦ Plane – 256 Sensors Vertical Array – f=2.25MHz.](image)

The analysis of Fig.3 shows that this process provides results in the fully saturated regime at distance 1, and partially saturated regime for all other distances, in the case of a vertical linear array. This is consistent with the presence of caustics at very short distance from the lens output face.

3.2. ACOUSTIC WAVE PROPAGATION.

Moreover, in order to anticipate for experimental results in terms of the shape of the acoustic field measured after propagation through a distorting surface, we developed a 3D propagation software based upon a parabolic equation calculating the propagated acoustic wave in an unbounded medium. Hence, we are able to observe the distorted acoustic field in various planes.
The refraction of the acoustic wave can be observed in Fig.4, especially in the vertical case (Fig.4 (a)), where caustics can be observed in good agreement with Fig.3 (a). Finally, we compared the coherence function calculated with the output of the simulation program presented in Fig.4 with experimental results of the same configuration.

Fig.5 displays the average coherence function along the linear array for the first propagation distance \( d_{src/rcvr} = 0.23 \, m \). The agreement between the two cases is quite good for the main lobe, and therefore for the radius of coherence, whose value can be related to the array gain [8].

4. CONCLUSION

The theoretical and simulation results presented here are used to define an experimental protocol leading to the acquisition of distorted acoustic signals in specific regimes of fluctuations predicted according to Flatté’s theory. The induced distortions in the acoustic field are characteristic of ocean perturbations, such as LIW. The effect of these perturbations on array gains is studied through the calculation of the coherence function. The latter displays a small radius of coherence both in experimental and simulation framework, which implies a degradation of the system performances.
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PASSIVE SONAR PERFORMANCE CHARACTERIZATION AND TRANSMISSION LOSS MEASUREMENT USING A CALIBRATED MOBILE ACOUSTIC SOURCE

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Abstract: A system and methodology for rigorously measuring the performance of a passive sonar detection system using a calibrated mobile acoustic source is presented. The source, an unmanned undersea vehicle (UUV) equipped with a projector of known directional and frequency response (omni-direction over the band 400 Hz-2 kHz), is capable of transmitting a pre-programmed waveform and dead-reckoning in accordance with a pre-programmed geometry. Through the use of a linear frequency-modulated (LFM) tracking sweep, a high-precision clock, and two GPS capable sonobuoys, the system employs a long baseline tracking algorithm to provide, real-time, ground truth reconstruction of the mobile source trajectory independent of the sensor under test. For a given passive acoustic receiver and broadband processor, it will be shown how performance metrics such as range-dependent probability of detection (P\textsubscript{D}) and false alarm rate (FAR) can be calculated concurrently with the measurement of in-situ transmission loss to quantify system performance and ultimately provide a measured figure-of-merit (FOM) for a desired system operating point. P\textsubscript{D} can also be decomposed as a function of signal-to-noise ratio (SNR) to provide a measurement of the system’s recognition differential (NRD). Data from a recent field test conducted on the New Jersey shelf break will be used to illustrate the methodology and interpret detection performance in the presence of an oceanographic anomaly, a cold pool duct, which was generated from the meander of a Gulf Stream eddy observed in August, 2011.

Keywords: Passive sonar, performance metrics, acoustic source, transmission loss measurement
1. INTRODUCTION

In this paper, a methodology is described for rigorously quantifying the performance of a passive sonar detection system using a calibrated, mobile acoustic source and clearly defined metrics. By using a precision-tracked source that supports independent ground-truth reconstruction, it is possible to reconcile concurrent measurements of transmission loss and detection performance to arrive at a measured figure-of-merit (FOM) for a passive sonar system without invoking theoretical assumptions typically used in order to link the system recognition differential (NRD) with a desired receiver operating point. Data from an experiment conducted off the New Jersey continental shelf in August, 2011 is used to illustrate the impact of diurnal changes in the ambient noise distribution on the FOM for an 8-element hydrophone array detection system.

2. METHOD

Central to the measurement methodology is the use of a calibrated, mobile source. The OASIS Mobile Acoustic Source, or OMAS [1], pictured in Figure 1, is a modified Expendable Mobile Acoustic Training Target (EMATT) originally developed for the U. S. Navy by LM Sippican. The EMATT has been modified by OASIS for use as a transmission loss (TL) measurement tool through the integration of a precision clock and calibrated acoustic source, and may be programmed to transmit specialized acoustic signals that have been optimized for tracking with a sonobuoy receiver baseline. Due to its mobility and very low self-noise signature, the OMAS represents an ideal surrogate target for detector performance characterization as the transmitted signal does not suffer from contamination by tow platform radiated noise. The vehicle is small (12.4 cm diameter, 91.4 cm long) and lightweight (10 kg), and is thus is easily launched by a single person as shown in Fig. 1(a). Prior to launch, the vehicle is programmed to run a series of defined tracks (e.g. opening, closing, circling, changing speed and/or depth) and the desired acoustic waveform is uploaded to the vehicle.

![Fig.1: OASIS Mobile Acoustic Source (OMAS): a) Deployment, b) OMAS source level measured in Dodge Pond for the Projector and vehicle radiated noise.](image-url)
Prior to use at-sea, calibration tests are conducted on each OMAS at the Dodge Pond Acoustic Test Facility in Niantic, CT, in order to measure the vehicle’s projector source level and directivity. This process is vital to the use of the OMAS for accurate TL measurements and quantification of detector performance. Figure 1(b) shows an example of the projector source levels measured as a function of frequency along with nominal levels provided by the manufacturer [1]. The source level peaks at about 152 dB re 1μPa, 1m at 1,100 Hz. The vehicle’s radiated noise spectrum is also typically measured while spinning a drag disk at the equivalent of 3 m/s (6 knots) advance speed. It is shown in Fig. 1(b) that the naturally radiated noise from the vehicle above 400 Hz is at least 50 dB below the levels generated by the acoustic projector. The spatial directivity of the source is also calibrated. The source is nearly omnidirectional in both the horizontal and vertical planes respectively for frequencies below about 1,500 Hz. Ping-to-ping variability for each unit is small (< 0.25 dB). Comparisons of one unit to another on the horizontal plane at broadside have shown repeatability in the mean to < 1 dB. For examples of representative OMAS spatial directivity response curves, the reader is referred to [1].

Accurate transmission loss measurements using the OMAS are made possible through the use of a precision clock in conjunction with careful time synchronization of the source and receivers. For TL and reconstruction, the receiver baseline is usually comprised of U. S. Navy DIFAR sonobuoys. The sonobuoys are deployed at one of several selectable depths, usually attached to a high flyer equipped with a GPS receiver to support precise position tracking. In addition to the programmed test signal designed to emulate a source of interest, the OMAS transmits a broadband linear frequency modulated (LFM) pulse that is received at each sonobuoy and matched-filter processed to determine the exact time delay between each signal’s known transmission time and its arrival at the receiver. Combined with the GPS position of each sonobuoy or receiver, this data may then be used to calculate the source range to very high precision on a ping-by-ping basis (up to several pings per minute). TL is then determined in real-time versus range and bearing from the difference between measured received level and the known calibrated source level [2].

### 3. RESULTS

We illustrate the use of the OMAS and the methodology for quantifying the performance of a passive sonar detection system with acoustic data from an August, 2011 experiment conducted in 250 m water depth near the North Atlantic shelf break, about 100 nmi off the coast of New Jersey, just south of the zonal shipping lane that accesses the Port of New York/New Jersey. The environment was downward refracting near the surface, but propagation was dominated by a cold pool duct that trapped sound between depths of 30 to 70 m, leading to extremely favourable propagation conditions. The sound speed profile (SSP) is shown in Figure 2.

Ambient noise in this location exhibited a strong diurnal dependence. Winds throughout the experiment were unusually calm for the time of year in the North Atlantic, so sea surface noise was not a dominant influence on the noise distribution. During the daytime hours, noise levels were generally driven by radiated shipping noise due to the proximity to the shipping lane. However, during night time hours an elevation in noise levels was routinely observed. This increase in noise was believed to be due to biologics, in particular, a species of croaker fish that are known to chorus in the 100-1 kHz band [3].
Fig. 2: Sound speed profile for the North Atlantic shelfbreak environment of interest (Source: N. Bogue, University of Washington Applied Physics Laboratory).

Fig. 3: Omni-averaged noise, beam noise, and array gain histograms at 758 Hz for shipping-limited and biologics-limited conditions.

Chorusing behaviour normally commences just after sunset and persists until dawn. Histograms depicted in Figure 3, show the (a) omni-averaged noise level in 2.3 Hz band, (b) beamformer output noise in 2.3 Hz band, and (c) array gain, measured at a frequency of 758 Hz. The array gain (AG) is defined as,
where $\sigma^2_n$ is the average noise power taken over all hydrophones in the sensor array, and $\sigma^2_b$ is the beamformer output power, for simplicity taken at broadside. In isotropic noise, the theoretical AG can be shown to be equal to the directivity index (DI), or $10\log_{10} N$, where $N$ is the number of sensors in the array [4]. From Fig. 3, we see that the during daytime, clutter-limited conditions, the mean noise level is about 56 dB in spectrum level, while at night, the noise level increases to about 62 dB in spectrum level. The difference in mean noise level during these two time periods has a direct impact on the detection performance of the passive sonar system under test, as will be quantified shortly. Notice the heavier tail content in the daytime AG histogram in Fig. 3(c). This behaviour is due to the capacity of the beamformer to spatially reject energy from the periodic passage of discrete interferers in the vicinity of the test. Contrast this to the AG histogram during croaker-limited time periods, which exhibits a very narrow distribution around a mean value of 10 dB. This value is very close to DI for the 8-element array used in the study, suggesting the spatial distribution of the croaker noise is very nearly isotropic.

The transmission loss measured in this environment is shown in Figure 4. The source and receiver are both in the duct. The TL shows a relatively flat (~10log r) range dependence between 1 and 10 km, consistent with the low loss duct supported by the cold pool intrusion at mid-water column depth evident from the SSP in Fig. 2. The histogram of the TL fluctuation about the mean, a measure of the uncertainty in the TL, exhibits a standard deviations of 4.5 dB. This number is somewhat high, most likely a consequence of how the test geometry sampled the spatial variability in the cold pool duct over time in this oceanographically dynamic Gulf Stream environment. Given the range dependence in the mean TL of Fig. 4, a 6 dB reduction in SNR due to the diurnal noise variation should translate into a nearly 4-fold decrease in detection range from day to night.

*Fig.4: Transmission loss vs. range as measured through the sensor under test at 758 Hz.*
The principal advantage of the OMAS tracking system described herein is the capacity to provide ground-truth reconstruction of the mobile target independently of the sensor under test. Independent target ground truth is essential for a truly objective and quantitative evaluation of a passive sonar system’s detection performance. The metric that most completely summarizes the detection performance of the system in a given TL and noise environment is the Probability of Detection vs. Range (PDR), where \( P_D \) is defined as that fraction of the time that the target is positively identified, when available to the sensor, at a given range. Detections opportunities are typically aggregated into third octave range bins for smoothing purposes. Measured PDR curves for an 8-element hydrophone array, deployed at mid-water column depth, corresponding to the noise distributions pictured in Fig. 3, are shown in Figure 5. To produce this plot, contact reports, or detection decisions, from the output of the passive sonar are compared to the target reconstruction. Detections that fall within an acceptable error tolerance in bearing and time are credited as correct, while detections that fall outside the tolerance window are penalized as false alarms. The gray histogram overlay in each plot indicates the degree of sample support in each range bin. The PDR curve of Fig. 5(a) corresponds to the daytime, clutter-limited case, while that of Fig. 5(b) corresponds to the nighttime, biologic-limited case. Ideally, the experiment geometry would be defined such that the PDR sweeps over its entire dynamic range, \([0, 1]\), but that is not always the case. We can clearly see from this result, however, the impact of the increase in noise level from day to night on the performance of the detector. The range at which the PDR crosses the 50% threshold, or \( R_{50} \), is a concise yet comprehensive way to quantify the performance of the detection system for a given set of environmental conditions. For the environment of interest and noise conditions described in Fig. 3, \( R_{50} \) decreases from 10 km under daytime noise conditions to a range of 2.5 km under night time conditions when the croakers are active. For the detection system under test in this example, the False Alarm Rate (FAR) is set to 0.

**Fig.5:** Measured Probability of Detection vs. Range (PDR) during (a) daytime, shipping-limited noise, and (b) night time, biologics-limited noise conditions, for zero false alarm rate. Samples are aggregated into third-octave range bins for smoothing purposes. The gray histogram overlay indicates the degree of sample support in each range bin.
Fig. 6: Measured Probability of Detection vs. Signal-to-Noise Ratio (PDS) at beamformer output during (a) daytime, shipping-limited noise, and (b) nighttime, biologics-limited noise conditions, for zero false alarm rate. Recognition differential may be inferred from the SNR at which the Hold Time crosses the 50% threshold.

The same data used to support the PDR curve can also be used to plot a Probability of Detection vs. signal-to-noise ratio (SNR) curve, or PDS, where SNR is defined at either the beamformer or element-level. The latter is best defined by correcting the beamformer output SNR back to the element level using the instantaneously measured AG. The PDS curve essentially normalizes out the environment dependence. However, it does have the advantage that it can be used to infer another commonly used detection performance metric, the Recognition Differential (NRD). NRD links the detection threshold to a desired receiver operating point. The associated PDS curves for the cases depicted in Fig. 5 are shown in Figure 6. The data are once again smoothed, this time by averaging into 1 dB SNR bins. Using this representation of detection performance, the NRD can be inferred by the SNR at which the PDS curve crosses the 50% threshold. Notice that for the system of interest here, the SNR$_{50}$ is equal to 10 dB. As expected, it is the same for both noise conditions as NRD is a scalar that describes the sensitivity of the detector and, thus, should be independent of the noise or transmission loss environment.

Typically, the NRD is theoretically linked to a desired (P$_D$, FAR) receiver operating point based on an assumption of Gaussianity [5], then employed in a sonar equation reconciliation to arrive at the Figure of Merit (FOM) for the detection system. The FOM is defined as the maximum transmission loss that can be tolerated by a system and still maintain the desired receiver operating point. It is best understood in the context of the passive sonar equation,

$$FOM = TL = SL - NL - NRD,$$

where SL is the source level, NL is noise level at the beamformer output, and NRD is the recognition differential, respectively. However, using the method described herein, instead of using an assumed NRD, the independent ground truth reconstruction provided by the OMAS system makes it possible to reconcile the measured TL in Fig. 4 with the measured R$_{50}$ determined from Fig. 5 to infer a FOM that is driven completely by data for a given source level and noise environment. In the case of the system under test in this example,
The $R_{50}$ values from Fig. 5 are 10 and 2.5 km, respectively, for daytime and night noise conditions. Reconciling these $R_{50}$ values with the daytime and night time TL measurements of Fig. 4 yields corresponding FOMs of 70.8 dB and 64.1 dB, respectively. As expected, the difference in day and night FOM agrees closely with the 6 dB difference in noise level associated with the evening onset of the chorusing biologics.

4. CONCLUSION

A system and methodology for rigorously measuring the performance of a passive sonar detection system is presented. Using a calibrated mobile target that supports in-situ measured TL and ground truth reconstruction independent of the system under test, it is shown to be possible to measure the detection range, NRD, or FOM of a detection system without invoking any mathematical assumptions of Gaussianity. There is a significant amount of information embedded in the measured PDR curve of a passive sonar detection system. As discussed in [6], while mean statistics of noise and TL reflect the macrostate of the ocean and the bottom environment, fluctuations in noise and TL relate to the environmental microstate, which can be critical for interpreting the uncertainty in detection performance. Future work aims to quantitatively link the standard deviation in TL, noise level, and even source level, with the slope of the PDR curve to arrive at more robust measures of detector performance, and ultimately improve performance prediction capability in temporally and spatially dynamic ocean acoustic environments.

5. ACKNOWLEDGEMENTS

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REFERENCES

MEASUREMENT AND MODEL FORECAST COMPARISON OF ACOUSTIC SIGNAL-EXCESS

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Abstract: In September 2012, mid-frequency acoustic signal excess measurements were conducted in the Gulf of Lions to study the impact of geophysical and water column environmental knowledge on sonar performance prediction. This paper presents a measurement and model comparison of the acoustic signal excess by using different categories of Rapid Environmental Assessment inputs. Specifically, five different categories of ocean environmental information have been used, including 1) climatology, 2) a high resolution regional ocean model, 3) a high resolution regional ocean model initialized by survey ship data collection, 4) high resolution regional ocean model updated by assimilating gliders’ data and 5) grey-ship equivalent on-scene for an operation. The bathymetric data used were either from climatology or collected by a multi-beam system for the scientific validation. The comparison was performed on the data collected along both range dependent and range independent tracks, seven source-target ranges for each track, two source depths at each source-target range and three different receiver depths. The impacts of varying degrees of fidelity of environmental information, as well as the conditions/configurations for setting up the acoustic model, on signal-excess predictions are summarized.

Keywords: acoustic signal excess, multi – static sonar system, acoustic model validation
1. INTRODUCTION

The Multistatic Tactical Planning Aid (MSTPA) is a tool currently in development at the NATO Science and Technology Organisation Centre for Maritime Research and Experimentation (CMRE). MSTPA is capable of modelling the performance of a given multistatic sensor network such as its ability to hold a potential target’s track. The acoustic model ARTEMIS (Adiabatic Reverberation and Target Echo Mode Incoherent Sum) implemented in MSTPA provides quantitative sonar performance predictions, e.g., the transmission loss, reverberation level at the sea surface and the sea bottom, and acoustic signal excess (SE). Among those outputs, SE is the one that has the most operational relevance because the probability of detection and the range of the day can be derived.

It is well established that the reliability of sonar performance predictions is limited by the knowledge of the environmental information and the associated uncertainty, and the validity of models used (including acoustic and oceanographic models). The objective of this work is to evaluate the sonar performance predictions of MSTPA using different categories of Rapid Environmental Assessment (REA) inputs such as climatology, forecasts with different data assimilation options, and in situ measurements. This work will lead to an understanding of the environmental information that is necessary to achieve the requisite (user defined) fidelity of an underwater acoustic prediction. Furthermore, this study also attempts to confirm the validity of the acoustic engine, ARTEMIS, and/or the necessary conditions/configurations for its correct operation.

2. THE EXPERIMENT

In September 2012, mid-frequency acoustic signal excess measurements were conducted during a REA phase of the Exercise Noble Mariner 2012 (NOMR12). Figure 1 depicts the bathymetry of the acoustic experimental site. The bathymetry data were collected by the shipborne multi-beam system during NOMR12. It is seen in Figure 1 that the bathymetry of tracks SE1 and SE3 are relatively benign whereas that of track SE2 varies abruptly. A top view of the acoustic SE experimental geometry is shown in Figure 2. There are seven acoustic stations along each track. The nominal distances of the stations are 0.1, 0.2, 0.5, 1.0, 3.0, 5.0 and 7.0 km from the VLA. The sensors on the acoustic VLA are unevenly spaced and span the middle of the water column. The acoustic target is moored at the depth of 81 m.

The acoustic source was deployed at two depths at each station. A sequence of acoustic pulses, with a broad band leading pulse and then a pulse formed by the summation of four CW (continuous wave) signals, was transmitted by the source repeatedly. This sequence of acoustic pulses was received by the acoustic target simulator, MERAS, which then re-transmitted the incident signal with a predetermined target strength and a programmable time delay.

At each acoustic station, one CTD (Conductivity-Temperature-Depth) cast was taken during the source transmission and used to provide the measured sound speed profile (SSP). Sediment cores were obtained along tracks SE1 and SE2 and then analysed in the laboratory to provide the acoustic sound speed and density at different locations along the tracks. The wave height of the sea surface and the wind speed were monitored continuously, by wave and metrological buoys throughout the entire acoustic experiment.
Figure 1: Bathymetry of the acoustic SE measurement area by the shipborne multi-beam system. SLIVA2: Acoustic vertical line array; MERAS: Acoustic target simulator; HYDRAS: Acoustic receiving arrays; SE107, SE207 and SE307: End points of three acoustic tracks (SE107 for track SE1, SE207 for track SE2 and SE307 for track SE3).

Figure 2: Top view of acoustic SE measurement geometry
MF Source: Mid frequency acoustic source; ScanFish: Sound speed profiler; OVC1: Oceanographic vertical chain; Wave Rider: instrument for measuring the wave height and direction; Metoc Buoy: metrological buoy
3. INPUT SPECIFICATIONS FOR MSTPA PREDICTION

Table I summarizes six groups of environmental inputs for running MSTPA cases. Column 1 (case 1) in Table I represents the least environmental knowledge and Column 6 (case 6) represents the most.

The oceanographic environmental data in cases 2, 3 and 4 are generated by the Regional Ocean Modelling System (ROMS) at CMRE. Specifically:
- ROMS-A: High-resolution ROMS output without assimilation
- ROMS-B: High-resolution ROMS output initialized from a CTD survey
- ROMS-C: High-resolution ROMS output with assimilation of glider data

To evaluate the effectiveness of the conventional knowledge, which represents that with the poorest resolutions both in time and space but would be available at the operation planning stage, MEDAR (Mediterranean Data Archaeology and Rescue) for the oceanographic environment and GEBCO (General Bathymetry Chart of the Oceans) 30” resolution for the bathymetry are used to form the first test case shown in Table 1.

Due to the fact that the pre-existing geoacoustic properties of the sea bottom were unavailable for this site, the average values of the sound speed and density derived from the sediment cores along the corresponding tracks are used for all six cases. The p-wave sound speed and density of the sea bottom were averaged over the values measured at 10 cm depth in the cores. The p-wave attenuation was set to 0.2 dB/wavelength.

For the work presented here, the standard Kirchoff method is adopted for calculating surface reflection loss, and the Chapman Harris formula is used for the surface scattering strength. Both values can be calculated if the wind speed is known. Additionally, bottom reflection is assumed to be Rayleigh reflection, and the bottom scattering strength is calculated according to Lambert's law. The Lambert coefficient $\mu$ is set to 0.00199526 (i.e. $10\log(\mu) = -27$ (dB)).

The monthly average of the wind speed is also not available for the experimental area. Hence, the average value of the wind speed obtained by the metrological buoy throughout the experiment period is used for cases 1 to 4, while the daily average of the measured wind speed is used for cases 5 and 6.

In order to eliminate the sonar system related parameters insofar as possible, the detection threshold in the sonar equation used to generate modelled data by MSTPA is set to zero. Hence, the actual quantity being compared is essentially the signal to noise ratio.

<table>
<thead>
<tr>
<th>Cases</th>
<th>1</th>
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<th>3</th>
<th>4</th>
<th>5</th>
<th>6</th>
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<td>Planning with oceanographic forecasts</td>
<td>Grey ship</td>
<td>Scientific validation</td>
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<td>ROMS-B</td>
<td>ROMS-C</td>
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<td>Average of measured</td>
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<tr>
<td>Wind speed</td>
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<td>Average of measured</td>
<td>Average of measured</td>
<td>Average of measured</td>
<td>Measured</td>
<td>Measured</td>
</tr>
</tbody>
</table>

Table 1: Environmental inputs for MSTPA predictions
In addition, in order to evaluate the necessary conditions/configurations for MSTPA’s correct operation, for each case listed in Table 1, two options for running MSTPA have been assessed:

- The smearing effect\[5\]
- The depth resolution

4. RESULTS

Tracks SE1 and SE2 are chosen for the comparison because they represent two extreme bathymetric conditions – benign and strongly range dependent. Further, the LFM (Linear Frequency Modulation) pulse and one of the CW pulses are used in the study as being representative of the typical signal types normally handled in MSTPA. The measured and modelled data are compared for seven acoustic stations on each track, with two source depths at each acoustic station, and three receiving depths for each source depth. The information regarding the experimental geometry is as follows:

- Distance from the source to the VLA: Calculated from the ship navigation data. Furthermore, the offset between the source position and the reference point for the ship navigation data was taken into account in the calculation
- Source depths at each station: 25 and 60 m
- Receiver depths: 24.77, 54.75 and 99.80 m

Figure 3 and Figure 4 show the selected model and data comparison results from this study. Furthermore, the performance was evaluated in terms of the root mean square of the differences between the measured and the modelled data. The environmental inputs that have the best performance in terms of predicting the SE (only for the LFM pulses used as an example) are summarized in Table 2.

5. CONCLUSIONS ON THE PRELIMINARY RESULTS

In terms of different sources of environmental data:

- For both SE1 and SE2 tracks, it seems that the SSP information is the dominant factor that affects the results, and there is no significant advantage of using the measured bathymetry.
- In situ measured SSP generally exhibits a better performance in predicting SE. Although cases 2, 4, 5 and 6 in Table I have similar performance for track SE2 when the source depth is at 25 m. However, it is safe to say that the ocean environment provided by MEDAR (case 1) has the worst performance for all the scenarios in this study. It suggests that a higher spatial and/or temporal resolution than the one provided by MEDAR is necessary for better SE prediction.
- The parameters (sea bottom p-wave attenuation and Lambert’s coefficient) chosen for this study seem to be sufficient for estimating bottom reflection loss and bottom scattering strength.
In terms of the setups for running MSTPA, the smearing effect should be taken into account, at least in this case study. A higher depth resolution is also apparently necessary in this case. However, higher depth or range resolution generally means a slower speed in producing the simulations. In order to run MSTPA adequately in real world scenarios, a rule of thumb for setting the range and depth resolution and using the different MSTPA options should be investigated.

![Table 2: Environmental inputs (case #) that have the best performance for the LFM pulses](image)

**Table 2**: Environmental inputs (case #) that have the best performance for the LFM pulses

<table>
<thead>
<tr>
<th>Track No.</th>
<th>Source depth (m)</th>
<th>Receiver depth (m)</th>
<th>24.77</th>
<th>54.75</th>
<th>99.80</th>
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<td>Measured SSP and Bathymetry (6)</td>
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<tr>
<td></td>
<td>60</td>
<td>Measured SSP and Bathymetry (6)</td>
<td>Measured SSP and Bathymetry (6)</td>
<td>Measured SSP and Bathymetry (6)</td>
<td></td>
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<tr>
<td>SE2</td>
<td>25</td>
<td>ROMS-C (4)</td>
<td>Measured SSP and Bathymetry (6)</td>
<td>Measured SSP and Bathymetry (6)</td>
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<tr>
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<td>60</td>
<td>ROMS-C (4)</td>
<td>ROMS-A (2)</td>
<td>ROMS-A (2)</td>
<td></td>
</tr>
</tbody>
</table>

**Figure 3**: Data/model comparison for source at depth of 25m and receiver at depth of 54.75 m for track SE1
Figure 4: Data/model comparison for source at depth of 25m and receiver at depth of 54.75 m for track SE2

6. ACKNOWLEDGEMENTS

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Session 16

Outer Continental Shelf, Shelfbreak and Canyon Acoustics

Organizer: Jim Lynch
A NORMAL MODE APPROACH TO MODELLING AIRGUN SIGNALS IN AUSTRALIAN COASTAL WATERS

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Abstract: A significant portion of the Australian offshore environment is characterised by sediments of significant elasticity. Moreover, thin and thick layers in the bottom can significantly affect the underwater sound propagation when considering the transmission loss of low to mid frequency signals. Seismic surveys conducted along Australia’s continental shelf are an opportunity to test shallow water sound propagation modelling results against measured data. The following work investigates the ability of adiabatic normal mode theory to model the transmission loss of broadband airgun signals over a range dependent bathymetry with a layered elastic bottom. The adiabatic mode results presented here are compared to a recent study, which considered both measured data and modelled transmission loss from a seismoacoustic parabolic equation model. Over a 2-100 Hz frequency band, the optimal sound transmission was observed around the modal cut-off frequencies of the sea surface-seafloor waveguide. At the lower end of the frequency band, propagation in this environment was dominated by one or two low order modes. The benefits and drawbacks of using an adiabatic normal mode approach are discussed in the context of both airgun signal modelling and the wider field of underwater sound propagation modelling. Ultimately, the aim of this work is to develop a better physical understanding of sound propagation in range dependent environments when the influence of elastic layers in the bottom cannot be ignored.

Keywords: Adiabatic Modes, Seismoacoustic PE, Seismic Surveys, Bass Strait
1. INTRODUCTION: THE AUSTRALIAN SHALLOW WATER ENVIRONMENT

The continental shelf along southern, western, and north-western Australia is characterised by an ocean bottom composed of calcareous sediments and calcareous rocks. The current sediment environment has been categorised as a temperate carbonate environment [1], the characteristics of which differ significantly from a tropical carbonate environment. The low depositional flux of terrigenous sediment along large sections of the Australian coast has exposed past sediments to weathering processes. Past mean sea level variations have reworked and cemented the shelf and slope carbonate sediments. This process has formed various cemented and semi-cemented layers of calcarenite that are present at the seafloor and within the sub-bottom [1].

Due to the cementation or partial cementation of calcarenite, it is appropriate to consider it as an elastic material. The shear wave speed in calcarenite has been estimated to vary from 500 m/s to 1400 m/s [2,3], depending on the degree of cementation and/or sand content. Since the average shear wave speed in calcarenite is lower than the sound speed in water, the major influence of the sub-bottom shear is to introduce attenuation into the propagating water column sound field. However, good sound propagation can occur at the modal cut-off frequencies of the ocean waveguide as discussed by [2] and [3].

1.1. BASS STRAIT AIRGUN DATA

Bass Strait is located between mainland southern Australia and Tasmania. It is also located within geographical area where layered calcarenite can occur. Acoustic measurements of an offshore seismic survey were recorded on hydrophone noise loggers located on the western edge of the continental shelf of Bass Strait. An in-depth summary of data collected is given by [2]. The location of the hydrophones and seismic survey sail transects are shown in Fig. 1.

![Fig. 1: Location and geometry of the array of sea noise recorders deployed in Bass Strait (with recorder’s/hydrophone’s numbers indicated). The blue and magenta lines show the location of the easternmost inshore and offshore seismic transects respectively.](image-url)
For this article, data from the inshore sail transect (blue line in Fig. 1) was considered along with the bathymetry profiles from the southernmost points on the respective transect to recorders 1 and 2. The airgun array of the seismic survey was approximately 7 m below the sea surface and the hydrophone loggers were anchored to the seafloor.

1.2. RANGE DEPENDENT ENVIRONMENTAL PARAMETERS

Two piece-wise linear bathymetry profiles were used to approximate the bathymetry from the source to receiver. The profiles that were used are shown in Fig. 2. The acoustic path over bathymetry 1 and bathymetry 2, respectively display moderate and strong range dependence. As such, a range dependent sound propagation model should be used to model the acoustic data collected from Bass Strait.

![Fig. 2: Piece-wise linear bathymetry profiles for range dependent modelling of signal propagation to noise recorders 1 (red) and 2 (blue).](image)

Range dependent acoustic environments with elastic bottoms have perviously been modelled with the parabolic equation (PE) method [4,5]. However, including both thin elastic layers in the sub-bottom and bathymetry is still a difficult senario to model. Some recent progress has been made on the problem; a seismoacoustic PE model described by [6] is capable of calculating the sound field considering range dependent elastic layers. This model was used by [2] to model transmission loss over the calcarenite bottom that is present in Bass Strait. However, the PE approach is very computationally intensive. As such, it is attractive to test and varify alternative modelling methods. The aim of this paper is therefore to explore the limits of using an adiabatic mode method to model sound propagation in a shallow water, range dependent environment with a calcarenite bottom.
2. THE GEOACOUSTIC MODEL FOR CALCARENITE

For the subsiquent sound propagation modelling, the geoacoustic parameters that were used for the seafloor are presented in Table 1. These were estimated from geotechnical data obtained from a 100 m borehole and are the same as discussed by [2].

<table>
<thead>
<tr>
<th></th>
<th>$H$ [m]</th>
<th>$\rho$ [g/cm$^3$]</th>
<th>$c_p$ [m/s]</th>
<th>$\alpha_p$ [dB/λ]</th>
<th>$c_s$ [m/s]</th>
<th>$\alpha_s$ [dB/λ]</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cemented Calcarenite Cap Layer</td>
<td>1</td>
<td>2.3</td>
<td>2600</td>
<td>0.2</td>
<td>1200</td>
<td>0.4</td>
</tr>
<tr>
<td>Slightly semi-cemented Calcarenite</td>
<td>100</td>
<td>1.9</td>
<td>2200</td>
<td>0.12</td>
<td>650</td>
<td>0.25</td>
</tr>
<tr>
<td>Semi-cemented Calcarenite</td>
<td>900</td>
<td>1.9</td>
<td>2100</td>
<td>0.12</td>
<td>550</td>
<td>0.25</td>
</tr>
<tr>
<td>Sedimentary Basement</td>
<td>N/A</td>
<td>3.0</td>
<td>3800</td>
<td>0.1</td>
<td>1900</td>
<td>0.2</td>
</tr>
</tbody>
</table>

Table 1: Geoacoustic parameters used for the sub-bottom calcarenite layered structure of Bass Strait.

A depth dependent sound velocity profile was also used. The sound velocity profile was calculated from the WOA 2008 seasonal climatology database and is shown in Fig. 3.
3. THE ADIABATIC MODE APPROXIMATION

Transmission loss was calculated using an adiabatic mode summation described by both [7] and [8]. The form used is given below in equation 1,

\[
TL(r, z) = -20 \log_{10} \left\{ \frac{1}{\sqrt{2\pi}} \sum_m \varphi_m(z, r) \varphi_m(z_s, 0) \frac{e^{i \int_0^r k_{r,m}(r') dr'}}{k_{r,m}(0) k_{r,m}(r) \int_0^r \frac{1}{k_{r,m}(r')} dr'} \right\}
\]

where \( TL \) is the transmission loss, \( \varphi_m(z, r) \) are the normalised mode functions, and \( k_{r,m}(r) \) are the modal wavenumbers. To evaluate the adiabatic mode sum along the acoustic paths of bathymetry 1 and bathymetry 2, the bathymetry profiles were first segmented. The normalised mode functions and modal wavenumbers were computed at each segment. The acoustic elastic normal mode program ORCA [9] was used to perform these calculations.

At each receiver depth the modal wavenumbers and mode shapes were interpolated to a finer computational range grid such that the range step was smaller than the acoustic wavelength. This was done to achieve a better spatial resolution of the sound field without having to run ORCA many times on a small range grid. The range integrals in equation 1 were then evaluated by trapezoidal integration. The transmission loss was calculated over a broad band of low frequencies, from 2 Hz to 100 Hz at a 1 Hz increment.

4. MODELLING RESULTS

Presented in Fig. 4 and Fig. 5 are the broadband transmission loss results for bathymetry 1 and bathymetry 2.
The data collected from the airgun array transects are shown as the black dots. For both bathymetry profiles the adiabatic mode transmission loss is in good agreement with the seismoacoustic PE. As discussed by [2], the low transmission loss peaks occur at the sea surface-seafloor cut-off frequencies due to low attenuating modes near the critical wavenumbers of the waveguide.

There are a few differences between the two numerical results, but in general the agreement is good. There is a greater difference between both numerical results and the measured data. However, the numerical results do agree with the trend observed in the data, implying a reasonable assumption for the geoacoustic parameters and bathymetry chosen. The remainder of this article will be focused on discussing the two numerical results, leaving further investigation of these data for future work.

5. DISCUSSION

For the shallow water environment with a layered calcarenite bottom modelled here, the influence of range dependence on the sound field can be considered as a parameterised band pass filter. As the bathymetry changes, so do the modal cut-off frequencies. Therefore, as a mode propagates in the ocean waveguide the amount of modal attenuation is both a function of bathymetry and frequency. This can be seen below in Fig. 6 which displays the variation of the modal attenuation for the first 5 modes for bathymetry 1 at a frequency of 23 Hz. The first range segment has a water depth is 105 m, and at 23 Hz mode 3 is close to its critical wavenumber. As the water depth increases there is an optimal depth where mode 3 propagates with the least amount of attenuation. The total sound field attenuation at any range is a given by the integrated variations of modal attenuation along the acoustic path. This determines the magnitude of the critical frequency transmission loss peaks and the frequencies at which the peaks occur.
There is good agreement when comparing the transmission loss from the two numerical methods, however adjacent to some of the main critical frequency peaks, the PE model predicts some small low loss peaks (see Fig. 4 and Fig. 5), whereas the adiabatic mode approach does not. These small peaks may be the result of coupling between modes close to the critical wavenumber. The transmission loss results from the two numerical methods differ by 1-2 dB with the adiabatic mode transmission loss slightly lower than the PE transmission loss. Since the difference is consistent across frequency, it may be attributed to the attenuating false bottom used in the PE model. This difference will be investigated as part of ongoing work.

Bathymetry 2 has stronger range dependant variations than bathymetry 1, but the adiabatic mode results are still in good agreement with PE results (see Fig. 5). Mode coupling may therefore be a secondary factor when modelling the sound field over a calcarenite style shallow water environment and over a band of low frequencies. This means that an adiabatic mode approach may be appropriate for more rapidly varying range dependent environments. While some further testing is required, the adiabatic mode method may be considered as a viable alternative to PE for these types of problems.

6. CONCLUSION

Transmission loss derived from airgun data from Bass Starit was compared to numerical results from an adiabatic mode method and from a seismoacoustic PE method. In general, for sound propagation modelling of environments where the sub-bottom is...
made up of layered low shear calcarenite, the adiabatic mode approximation is a reasonably accurate method. Similar to the range independent case, the sound field is dominated by modes near their critical wavenumber. For modal wavenumbers near the critical wavenumber, variations in bathymetry can result in an optimal propagation depth where modal attenuation is minimised. For strong range dependent bathymetry the adiabatic mode results are consistent with PE results and therefore may be considered as an alternative numerical modelling approach for shallow water calcarenite type environments.

7. ACKNOWLEDGEMENTS

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REFERENCES


OBSERVATIONS OF HORIZONTAL COUPLING IN THE MONTEREY BAY CANYON

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Abstract: Data has been collected on acoustic vector sensors mounted on autonomous underwater gliders in the Monterey Bay during 2012-2013. In this work, we show results of intensity processing to estimate bearing to impulsive sources of interest. These sources included small explosive shots deployed by local fisherman, and humpback whale vocalizations. While the highly impulsive shot data produced unambiguous bearing estimations, the longer duration whale vocalizations showed a fairly wide spread in bearing. The causes of the ambiguity in bearing estimation are investigated in the context of the highly variable bathymetry of the Monterey Bay Canyon, as well as the coherent multipath interference in the longer duration calls.

Keywords: three-dimensional propagation, vector intensity, bearing estimation
1. BACKGROUND

Applications using vector sensors and vector field processing have increased tremendously over the past decade or so. In addition to new measurement technologies, various approaches for processing vector sensor data have been developed and applied to both single sensors and arrays of sensors. Fundamentally, there are two approaches for processing data collected from vector sensors: additive, whereby the acoustic pressure and components of acoustic particle velocity are summed together coherently; and multiplicative, whereby the pressure and the components of velocity are multiplied together to form the acoustic intensity vector. These approaches, and variants of them, have been studied extensively by numerous researchers (e.g., Hines, et al. [1], and Cray and Nuttall [2]).

Some of the earliest studies of the fundamental structure of the acoustic vector field were performed by Mann, et al. [3], in which the formal definitions of the active and reactive components of the acoustic intensity vector were defined. More recent work explored new aspects of the acoustic intensity field, as summarized by Smith [4].

Due to the growing interest in the acoustic vector field, various researchers were motivated to adapt existing numerical propagation models to compute the associated particle velocity field. The first reported adaptation of a parabolic equation model was done by Smith, et al. [5], and formal definitions of the equations to be invoked in both parabolic equation models and normal mode models were later developed by Smith [6], [7]. In this latter work, Smith was the first to theoretically establish that the particle motion in a waveguide characterized by normal modes is elliptical [6]. This was later illustrated numerically by Dall’Osto and Dahl [8]. A consequence of this is that the acoustic intensity vector rotates when multipath interference is present, in contrast to the unique intensity vector direction due to passage of a plane wave or uniquely resolved multipath arrival.

In March 2012 and September 2013, the Naval Postgraduate School deployed an autonomous underwater glider with an integrated acoustic vector sensor for observations of ambient noise conditions in the Monterey Bay area of California. There were numerous broadband signals collected, including boat noise, marine mammal vocalizations, and even several impulsive signals transmitted from “seal bombs” used by local fishermen to discourage harbor seals and sea lions from taking their catch, hereafter referred to as “shots”.

Upshaw [9] processed numerous signals using various approaches in order to determine bearing from the glider to the noise source. Here we shall focus on the results from the directional intensity processing, which showed some interesting behaviour for different noise sources. Specifically, he examined the short duration shots recorded in 2012 as well as some humpback whale vocalizations recorded in 2013. An obvious difference in the signals analysed can be seen by examining their respective spectrograms, as presented in Fig. 1. The spectrogram of a string of 5 whale vocalizations displayed in the left panel shows that the signals had a bandwidth of around 75-150Hz, a duration of around 1sec, and the signal-to-noise ratio was only about 10dB at best. In contrast, the spectrogram of 2 shots displayed in the right panel shows very broadband signals, covering the entire processing band used (350-650Hz), with a very short duration of around 50msec and large signal-to-noise ratios over the entire band of around 30dB.

In order to examine the vector intensity bearing estimation, a bubble plot was created as a function of time that depicted the relative amplitude of the intensity vector as...
proportional to the size of the bubble, centered at the true bearing of the signal relative to the glider’s position. The results of these calculations are depicted in Fig. 2: the left panel displays the results for the 5 whale vocalizations, and the right panel displays the results for the 2 shots.

Fig.1: Pressure spectrograms of impulsive signals measured during deployments of underwater gliders in Monterey Bay; five humpback whale vocalizations (left panel) and two explosive shots (right panel).

Fig.2: Intensity vector bearing response. Bubble size indicates magnitude of intensity vector; five humpback whale vocalizations (left panel) and two explosive shots (right panel).
As can be observed in Fig. 2, the bearing estimation to the whale vocalizations showed significant ambiguity, generally displaying a 40deg or larger spread in direction. The shot data, on the other hand, showed unambiguous bearing estimations with a spread of less than 10deg. While the lower SNR of the whale vocalizations could be a factor, close examination of the bearing ambiguity showed that the evolution of the intensity vector was not random, but rather seemed to sweep continuously over the range of bearing uncertainty. This suggested that it could be something fundamental in the propagation of the signals.

Finally, upon displaying the locations of the glider at the time of the data recordings, and illustrating the general directions towards the noise sources, as depicted in Fig. 3, it was noted that there was a significant difference in the bathymetry along the propagation paths. The whale vocalizations were transmitted across the highly variable bathymetry of the Monterey Bay Canyon, while the shots were transmitted across a relatively benign shelf region north of the canyon. It is the impact of this 3-dimensional variable bathymetry, possibly coupled with the narrower band and longer duration signals, that are potential causes of the horizontal bearing ambiguity, and which we will investigate in this paper.

Fig. 3: Geospatial direction of intensity vector bearing results relative to glider position in Monterey Bay; humpback whale vocalizations recorded over canyon (left panel) and explosive shots recorded over shelf (right panel).

2. NUMERICAL MODELING RESULTS

In order to compute complex intensity vector pulse arrivals in such a highly 3-D variable environment, a broadband 3-D version of the Monterey-Miami Parabolic Equation (MMPE) model was employed (Smith [10]). Similar versions of this model have been employed to compute the interaction of the acoustic vector field with non-linear internal waves by Dossot, et al. (e.g., [11] and [12]). For this work, the model was
adapted to allow 3-D variable bathymetry, while the sound speed profile and geoacoustic properties were defined independent of range and bearing. The bottom was assumed to be a fluid-like homogeneous half-space with sound speed 1700m/s, density 1.8g/cm³, and attenuation of 0.2dB/m/kHz. The ocean surface was assumed to act as a flat, pressure release boundary. The sound speed profile in the water column was based on average profiles measured during the course of the Sept. 2013 experiment, and is depicted in Fig. 4 (left panel). The bathymetry for the Monterey Bay Canyon was obtained from the Southern California Coastal Ocean Observing System [13], and gridded at roughly (1/20)min in latitude and longitude. The region of the canyon where the whale call vocalizations were recorded was extracted for calculations, as depicted in Fig. 4 (right panel).

In order to estimate the broadband, impulsive arrival structure, the 3-D MMPE model was run for 512 frequencies over a bandwidth of 127.5Hz centered at 400Hz. This resulted in a frequency bin size of 0.5Hz, giving a total time window of 4sec. The source depth was 92m, consistent with the depth of the glider at the time of the whale vocalizations being examined. While this time window is just long enough to capture the primary structure of the measured impulse response, the smaller bandwidth was chosen simply to reduce computational run time. Future work will be performed using more processors over a larger bandwidth.

The 3-D MMPE model employed for this work computes both the acoustic pressure and associated acoustic particle velocity, as described by Smith [7]. At the time of this writing, only processing of the complex pressure field was completed. However, there is enough structure in these results for us to begin making some statements of the expected behaviour of the predicted vector intensity field.

The time domain response of the field as a function of cross-range distance will provide us a view of the existence of horizontal multipath structure. As can be seen in the plot of bathymetry, a large slope exists for positive cross-ranges near the source. Once the forward range exceeds about 3km, there are large slopes on either side of the forward propagation. This is consistent with observations of the time domain response at a fixed depth, as depicted in Fig. 5. Specifically, near the source at a forward range of 100m, only the direct and, near the central radial, surface reflected path are observable. At a forward range of 1km, the bottom slope at positive cross-ranges has started introducing diffuse scattered energy. At 2.5km, there is significant scatter for positive cross-ranges, and
bottom scatter from negative cross-ranges can also be seen to enter the image. Finally, at 5km, there are multiple regions of scattered energy for positive cross-ranges.

What is particularly interesting in these plots is the multipath structure which appears to exhibit different angles of arrival. This would suggest the existence of horizontal coupling. In order to confirm this, small segments (extending 400m over 128 cross-range points) were extracted from the solution at 5km. This data would mimic what might be observed on a horizontal array towed parallel to cross-range at 5km, as displayed above in Fig. 5(d), for which we can apply plane-wave beamforming techniques.

In Fig. 6, results are shown for such plane-wave beamforming for three different array locations. Specifically, the data is extracted corresponding to an array center at 0km, and +/-1km in the cross-range direction. For the 0km array center results in Fig. 6(a), the arrival structure is dominated by a strong arrival at broadside (0 deg), corresponding to the direct path. The arrival structure following that is more diffuse, but is still concentrated near broadside. However, weaker subsequent arrivals are seen to occur at negative angles on the array (going from positive to negative cross-ranges), as is consistent with the general features that can be observed in Fig. 5(d). Finally, there is a weak arrival at around +50 degrees, indicating a large horizontal scattering feature to the left of the source. This feature, though, is significantly lower in intensity than the other arrivals structures, and should be considered with caution as numerical noise may begin to play a role at these levels.

Fig.5: Plots of travel time versus horizontal cross-range at a fixed depth of 100m for varying distances forward from the source: (a) range = 100m; (b) range = 1km; (c) range = 2.5km; (d) range = 5km.
For the arrays centered at +/-1km, as depicted in Figs. 6(b) and 6(c), the behaviour is similar. In addition, it is worth noting that the earliest arrivals appear at roughly +/-15deg, consistent with the offset of the center of the array axis, the strong secondary arrivals both tend to shift towards negative angles, consistent with the earliest bottom reflections to the right of the source (at positive cross-ranges) generating horizontal scattering towards the left (towards negative cross-ranges). In both cases, the late weak arrival appears at a large positive angle, indicating strong horizontal scattering from the left. As before, though, interpretation of these last arrivals should be treated with caution.

**Fig. 6:** Plots of travel time versus horizontal arrival angle at a range of 5km and depth of 100m for a 400m array centered at various cross-range position: (a) array center = 0km; (b) array center = -1km; (c) array center = +1km.

3. **SUMMARY**

This work was motivated by the potential impact of horizontal coupling on the accuracy of bearing estimation from measurements of the vector field. While this preliminary work has not yet established a direct, quantifiable impact, it is clear that the nature of the bathymetry in canyons can introduce significant out of plane scattering of the acoustic propagation. Work is on-going to evaluate the corresponding structure in the vector field, and the impact of other parameters such as temporal coherence and ambient noise.
4. ACKNOWLEDGEMENTS

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The “integrated ocean dynamics and acoustics” (IODA) hybrid modeling effort


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Abstract: Regional ocean models have long been integrated with acoustic propagation and scattering models, including work in the 1990s by Robinson and Lee. However, the dynamics in these models has been not inclusive enough to represent submesoscale features that are now known to be very important acoustically. The features include internal waves, thermohaline intrusions, and details of fronts. In practice, regional models predict internal tides at many locations, but the nonlinear steepening of these waves and their conversion to short nonlinear waves is often improperly modeled, because computationally prohibitive nonhydrostatic pressure is needed. To include the small-scale internal waves of tidal origin, a nested hybrid model is under development. The approach is to extract long-wavelength internal tide wave information from tidally forced regional models, use ray methods or mapping methods to determine internal-tide propagation patterns, and then solve two-dimensional high-resolution nonhydrostatic wave models to “fill-in” the internal wave details. The resulting predicted three-dimensional environment is then input to a fully three-dimensional parabolic equation acoustic code. The output from the nested ocean model, run in hindcast mode, is to be compared to field data from the Shallow Water 2006 (SW06) experiment to test and ground truth purposes.

Keywords: acoustic propagation modelling, internal tides, nonlinear internal waves, ocean dynamical modelling
1. INTRODUCTION

The computational modelling of ocean water movement and ocean water property evolution has a history spanning more than 50 years. The rationale for this effort is multifaceted, including prediction, real-time calculation of actual conditions with wide dynamic range, and idealized modelling in order to foster understanding of the physical processes at work. The processes range in scale from climatic to wave-period in duration, and from planetary to pebble-sized in length scale. Here, we present work at medium scales, aimed at features that have been determined to be of interest from an underwater acoustic propagation standpoint, of hours to weeks in duration, and tens of meters to tens of kilometres in length scale.

This work involves the interfacing of so-called regional ocean models, running in data assimilation (DA) mode and covering spatial scales from hundreds of kilometres down to about a kilometre, to models suited to smaller-scale features. In DA mode the model outputs are driven towards reality as much as possible, being typically data-starved by nature [1-2]. Modern data-driven regional models use the hydrostatic pressure assumption, where water vertical acceleration is not fully considered. (Vertical force does not equal mass times acceleration.) Including the nonhydrostatic pressure makes the computation far more intensive.

The reason for interfacing these models is that, through recent research efforts, the community has come to realize that nonlinear internal waves (NIW) of order 100-m wavelength exhibiting nonhydrostatic pressure physics can have profound impacts on sound propagation. These waves can focus energy [3-6], and can affect coherence [7-8], multipath interference [9], and reverberation [10]. However, these waves are not properly handled by the hydrostatic regional models, so their properties cannot be predicted simply by extending the data-driven regional modelling approach to smaller scales and periods.

Here, a hybrid (or composite) method for computing highly resolved three-dimensional waves with nonhydrostatic pressure (NP) dynamics within ocean volumes is outlined, where only the longer-wavelength features are modelled in data-driven fashion. The method is aimed at the prediction of NIW that develop from long-wavelength internal waves at tidal frequency (internal tides, IT), which arise in the regional models. The method has not yet been proven to be sufficiently accurate to improve activities like sonar system performance prediction. Making this assessment is one of our research goals. The development and testing of this hybrid model is one component of a larger project, IODA, defined in the title, which also includes basic research in IT formation and dynamics, NP wave dynamics, surface wave modelling, regional model development, statistical and computational acoustic propagation modelling, NP computational modelling, and efficient or optimal interfacing of ocean environment models and acoustic models (i.e. passing sufficiently but not unnecessarily detailed sound-speed structures to acoustic models).

2. THE HYBRID MODEL COMPONENTS

The first component is a regional model (RM) running in DA mode with tidal forcing, so that internal tides develop. The mechanism for IT generation is oscillatory flow near a sloped seabed, so that boundary conditions induce oscillatory vertical flow [11]. In deeper waters, IT tend to have small current and displacement amplitudes, thus are linear, and are well approximated as exhibiting hydrostatic pressure (HP) dynamics. In addition, as the IT
move toward shallow water, they can be described by only a few vertical baroclinic flow normal modes.

This leads to the **second component**: a ray-tracing model describing the propagation of IT normal modes. This model requires the modal properties to be computed from the de-tided output of the regional model. Required are vertical structures of internal-wave modes (eigenfunctions), with phase speed $(c)$ related to the eigenvalues. It also requires IT initial conditions from the regional model. This component gives IT properties in an area of interest. In some situations, it may be possible to extract the IT properties from the model using filters, but in general it is difficult to separate the IT field from eddy features, particularly moving eddies. This exposes an important caveat for the ray-trace component: The background through which the IT waves propagate does not vary slowly compared to the waves and at scales that are larger than the waves, so ray tracing is not rigorously valid. But it is nonetheless done. The validity of this can be examined by comparing full RM output with de-IT-processed RM output plus traced IT fields. Typically, rays begin where IT waves are found to form (or believed to form).

As stated, the NIW evolve from the internal tides as they propagate [12-13]. This is modelled with the **third component**, a nonlinear wave evolution model with NP dynamics. The most familiar model is the two-dimensional (vertical slice) Korteweg-de Vries model. Here, we use a generalization of KdV, the extended KdV model with earth rotation [14], which we abbreviate as eKdVf. This is a cubic nonlinear equation for internal-wave mode amplitudes in a time-space domain. Writing mode-one amplitude as $\eta$, the equation is

\[
\frac{\partial}{\partial s} \left( \frac{\partial \eta}{\partial t} \right) + (c + \alpha \eta + \alpha, \eta^2) \frac{\partial \eta}{\partial s} + \beta \frac{\partial^3 \eta}{\partial s^3} - \frac{c}{Q} \frac{dQ}{ds} \eta = \frac{f^2}{2c} \eta
\]

The spatial dimension $s$ is along-ray distance in our scenario. The nonlinear term coefficients ($\alpha$’s) and the NP-dispersion coefficient ($\beta$) are specific to the mode being worked on; they are integrals involving the mode eigenfunctions. $Q$ is a related variable that allows slowly-varying depth. Rotation effects are imparted by $f$, equal to 2 times latitude times the earth rotation rate. The eKdVf is solved along IT rays to give short-wavelength NIW along the rays. We solve only along mode-one IT rays because mode one dominates on continental shelves [13], our major interest. The along-ray mode-amplitudes $\eta(s,t)$ are then converted to time-dependent 2D along-ray sound-speed fields, and then interpolated to give the evolving 3D sound speed field. The eKdVf solution is a refinement of the IT field present in the RM, and of the IT field calculated with ray tracing, so these are not passed on to the fourth and final component.

The **fourth component** of the model is a 3D parabolic equation sound propagation code [15]. This is run sequentially in time to produce evolving sonic fields, which we call 3.5D modelling (as opposed to true 4D modelling of sound in moving water).

### 3. DEMONSTRATION OF SIMULATION OUTPUT

The modelling is perhaps better understood by viewing examples of the outputs. Figure 1 shows fields from the data-assimilative RM [16], developed for real-time predictions in the ONR SW06 experiment, and since refined (reanalysis: improved initial conditions and data; improved resolution). The surface temperature variations (left) illustrate the internal-wave mode speed anomalies and circulation features that the IT, visualized at right, must propagate through. The internal tides are analyzed by projecting vertical profiles of press-
Fig. 1: (left) Surface temperature snapshot from the MIT MSEAS data-assimilative regional model, summer 2006 conditions. (right) A snapshot of 30-m depth semidiurnal-band filtered temperature fluctuations, visualizing ITs that propagate away from the continental-shelf edge (closely bunched contours) in both directions. Initial conditions for an along-ray eKdVf calculation will be similar to a sequence of values of this parameter at the ray origin. IT interference can be seen. The IT in this model propagate through and interact with the complex environment seen at the left, as do ray-traced IT for the hybrid model.

Fig. 2: (left) The mode-1 long wave (asymptotic) internal wave speed for the background 3D model flow field is shown, in color, for one particular wave direction. Mode-1 internal-tide rays starting at a line of hypothetical sources are plotted in black. Two isophase fronts are superimposed in white. (right) Areas of surface current convergence for full-bandwidth nonlinear internal waves simulated with eKdVf are shown for two scenarios. The waves computed with advection taken into account are shown in red; these use the rays shown in the left panel. The waves computed with the background current set to zero are shown in blue.
ure anomaly and/or velocity onto the vertical dynamical normal mode shapes, then Fourier transforming in 62-hour windows to obtain time series records of normal-mode oscillation amplitudes at the resolved frequencies. Oscillations near the diurnal and semidiurnal forcing frequencies dominate. Fig. 1 (right) does not show this analyzed product, but instead simply shows a snapshot of semidiurnal band-passed temperature fluctuations at 30 m. This looks very much like the mode-1 signal because mode-1 is dominant.

Next, the left panel of Fig. 2 shows results of internal-tide ray tracing. Rays are initialized at locations along a line near the critical slope location, which often divides on-shelf IT energy flux from toward-ocean flux. At the critical slope location, semidiurnal internal-wave 3D characteristics, in the horizontal direction parallel with the bathymetric gradient, are parallel with the seabed. Critical locations often run along a line, so the example internal-tide rays start along a line. On the left, the rays are shown, including the effects of advection by sheared currents (functions of $x, y,$ and $z$) extracted from the model. The IT rays that are shown result from anisotropic ray tracing, using wave speed derivatives in $x, y$ and azimuth (ray direction), which are found by solving the $k=0$ (long-wave) version of the Taylor-Goldstein equation.

On the right of Fig. 2 are snapshots of simulated nonlinear internal waves (mode-one amplitudes) computed by first solving the evolution equation along each ray, then interpolating to fill areas between the rays. Initial conditions are IT sine waves, identical for all rays. Two sets of waves are shown, one computed along IT rays that include advection by currents, and one along rays that ignore current and are controlled only by $(x, y)$ gradients of stratification and water depth. The dark traces show areas of strong surface convergence in waves computed by eKdVf; these are the bright streaks in typical satellite synthetic radar images of internal waves. The currents are seen to push the southern waves to the southwest. A sharp bend appears in the predicted wavefront with or without currents. Importantly, initial conditions in this demonstration are the same for each ray, and are not taken from the RM.

The final step is to feed the internal wave fields to the acoustic PE model. Here, 200 Hz sound is propagated about 8.5 km. The sound approaches the waves from ahead (from the west) then enters the waves. To feed the wave information to the PE, the 2D-interpolated mode-one amplitudes (Fig. 2, right) are first bilinearly interpolated onto a 40 by 40 meter square grid. Then, the demonstration shown here uses a locally representative mode-one shape to build a 3D displacement field, from which a 3D sound-speed field is computed by adjusting depths of a background sound-speed profile, computed from background temperature and salinity profiles. The full model, in development, will use the spatially-dependent background conditions (already used in the wave evolution model) to estimate mode shapes everywhere, and sound-speed profiles everywhere, refining the results. Fig. 3 shows, at the left, the domain where 3D sound-speed is computed, and the acoustic domain. A horizontal planar slice through 3D acoustic field is shown at the right. The PE domain covers only a small portion of the area of the internal-wave simulation (Fig. 3, left).

It is immediately evident from Fig. 3 (right) that the sound propagation has an axisymmetric nature near the source, with a ring pattern of intensity at this depth, explainable by mode interference. Within the internal waves, mode coupling occurs, with mode coupling details being a function of position along the line where the sound enters the waves (the left edge of the waves). This is because the acoustic-source to wave distance varies with the azimuth of these lines, varying the relative mode phases [17].
4. INTRIGUING MODELING ISSUES

The modelling that we have accomplished to date involves a number of approximations, or simplifying assumptions, that need to be validated. (Or abandoned, if necessary.) Chief among these is the consideration of only mode-one internal tides while setting up and running the eKdVf prediction of nonlinear internal waves. The use of mode-one eKdVf solutions to simulate the high-frequency waves is justifiable based on similarity of modelled and observed fields. On the contrary, satellite SAR images suggest that nonlinear waves on the shelf have more complicated spatial patterns than the mode-one ITs that the hydrostatic RM predict. This is because many other internal waves are present in the real ocean which are not resolved nor initialized in the RM (due to lack of measurements). Mode one dynamics along may thus be insufficient for eKdVf initial conditions along rays (not yet implemented in the example, to be clear).

At this time, we plan to use mode-one IT amplitudes along the critical-slope line (looking something like an along-slope isobath) to define likely IT ray starting points, and use IT phase gradients to define ray initial directions. However, field studies [18] and computational studies [19] have shown that internal tides can move upward in beams from the seafloor into the thermocline, with beams being equivalent to multiple modes. Uncoupled normal-mode propagation is a simplification of what actually happens when long-waves of many modes propagate into shallow water, increasing in energy density to the point where nonlinear effects are in play [20]. Because the energy density in the main thermocline must be a determining factor in the generation of thermocline-trapped (mode-one) high-frequency waves, the modal interference behaviour must be relevant. For these reasons, we are considering ways to use a multi-mode model to devise input conditions for the eKdVf. This may be consistent with taking initial condition directly from the RM thermocline dynamics (thus, in addition to using detided RM fields to compute the back-

Fig. 3: (left) The area of the Cartesian 3D PE acoustic computation through the internal wave field computed with advection (Taylor-Goldstein, blue in Fig. 2) is shown with a white box. The red arrow indicates that the sound source is at the center of the left box edge; the arrow points in the marching direction. The black box shows interpolated area available for the 3D sound-speed calculation. (right) A snapshot of transmission loss at 32.25 m depth, 200 Hz is plotted. A transition occurs as sound energy enters the waves. The water depth is 80 m at this site, and the internal waves are 20 to 25 m in height.
ground medium for eKdVf-deduced shelf internal tide and waves, using the residual fields for internal-wave initial conditions.). However, the modal content of internal tides may not be well described by the hydrostatic RM with limited bathymetric resolution, so an accurate interference pattern may be challenging to extract.

Another concern is the energy of the nonlinear internal wave field. The waves depicted in Figs. 2 and 3 have higher energies than seen in the field (not shown). The eKdVf model has no dissipation, whereas short nonlinear waves are well known to spawn turbulence and to dissipate as they travel. An empirically-tested wave dissipation scheme may improve the comparison with actual waves. Another possible cause of excess wave energy is the fact that initial conditions seem to cause localized “point” sources of oceanic nonlinear waves, while the scheme presented here does not account for the cylindrical spreading loss that would result. Our current scheme would need an additional inter-ray distance scaling of wave energy to account for spreading loss. This issue of apparent point sources of nonlinear internal waves ties in with the issue of initial condition variability for the mode-one eKdVf along the locus of ray origin points.

5. SUMMARY

The linked ocean mesoscale, internal wave, and acoustic modelling effort is described here, and is demonstrated with a small amount of output. An example computation illustrates the potential of the method for localizing internal waves, and for computing their acoustic effects, in regions spanning thousands of square kilometres. The outstanding technical issues are also discussed, generally regarding the modelling of short wavelength internal waves.

6. ACKNOWLEDGEMENTS

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Session 17
Radiated Noise from Ships and Production Platforms

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ANALYSIS OF UNDERWATER ACOUSTIC NOISE MEASURED AT THE SHIP BOW DURING SEA TRIALS

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Abstract: Measurements of underwater acoustic noise radiated by ships are typically performed by an experimental set-up positioned at a distance by the target. To determine the radiated noise level, at 1 meter from the ship acoustic center, the transmission loss should be computed and great attention should be paid in setting the parameters of the adopted propagation model. The chance to assess the underwater noise at the ship bow represents a great opportunity to investigate the radiated signals in an alternative way, working in a particular location, at short distance from the acoustic sources of the vessel, whatever and wherever they are. In addition, this measurement allows to determine the sonar self-noise. This paper presents an analysis of the self-noise observed at the bow of a modern ship during sea trials, by the means of both broadband levels magnitude and spectral analysis. The self-noise has been obtained by selecting specific records among a large set of noise measurements with the aim of evaluating a number of different situations in which the ship can operate. In particular, it has been analysed the effect on the noise spectrum structure of the propulsion mode (in terms of auxiliary machineries activity status and propellers rotational speed) and of the ship conduct (in terms of speed and heading). Nevertheless a widespread and heterogeneous set of acoustic sources influences the measure of noise, some interesting remarks regarding the relation between the self-noise and the above-mentioned factors are proposed.

Keywords: Ship noise spectrum, sonar self-noise, underwater noise measurements
1. INTRODUCTION

The measurement of underwater acoustic noise radiated by a ship is a task as important as difficult to accomplish for a number of reasons.

Since world war II the acoustic signature of military ships has been measured and analysed [1,2] and in the last two decades several measurements of radiated noise by commercial vessels have been proposed in the literature [3-6]. Moreover, the underwater noise pollution has recently acquired attention by national and international bodies that are currently involved in the formulation of norms and requirements for the shipping field [7].

Measurements of underwater acoustic noise radiated by ships are typically performed by an experimental set-up positioned at a distance by the target [7,8], by observing the ambient noise level variation during a ship passage. Then, in order to evaluate the noise source level, the transmission loss is computed with great care [3,6] by considering the environmental conditions and attention should be paid in setting the parameters of the adopted propagation model [9,10].

In this paper we present the analysis of noise measurements at the bow of a ship. This particular location has allowed the noise to be measured continuously during sea trials, allowing to precisely correlate the level magnitude with the actual navigation parameters, with the aim of evaluating a number of different conditions in which the ship can operate. In particular, it has been analysed the effect on the noise spectrum structure of the propulsion mode (in terms of auxiliary machineries activity status and propellers rotational speed) and of the ship conduct (in terms of speed and heading). Moreover, it has been evaluated the sonar self-noise level for a set of frequency bands, by selecting specific records among a large set of noise measurements. Nevertheless a widespread and heterogeneous set of acoustic sources influences the measure of noise, some interesting remarks regarding the relation between the self-noise and the above-mentioned factors are proposed.

This paper is organized as follows. In Section 2 the measurements set-up is described together with the description of the considered sea trial and the results of the ship noise analysis. Finally, in Section 3, the conclusions are drawn.

2. SHIP NOISE ANALYSIS

The noise measurements were performed at the bow of a modern ship during sea trials; this permits to obtain noise sequences for a number of different operative conditions. The noise signal has been acquired by an omnidirectional hydrophone and registered in contiguous records.

Selected records of noise, having length from 60 s to 120 s, have been processed to obtain a noise spectral level (NSL) with 1 Hz resolution measured in dB re 1 $\mu$Pa$^2$/Hz. The NSLs in 1 Hz band have been analysed in octave bands. The self-noise level has been computed by integrating the noise spectrum in 1 Hz band over three different frequency bands. The frequency bands are representative for low, medium and high frequencies of the noise spectrum: LF band, from 1 to 100 Hz; MF band, from 100 to 500 Hz; HF band, from 500 Hz to 4 kHz.

The noise records considered in this paper was acquired during a sea trial characterized by ship’s speed variation (relative to a reference speed value, higher than 5 kn) and heading values as shown in Fig. 1.
Fig. 1: Values of speed variation (left) and heading (right) during the sea trial. The values correspond at the start (continuous line) and at the end (dashed line) of each record.

<table>
<thead>
<tr>
<th>Group</th>
<th>Turning</th>
<th>Speed variation</th>
<th>Stable condition</th>
<th>All aux active</th>
</tr>
</thead>
<tbody>
<tr>
<td>Records [first, last]</td>
<td>[8, 10]</td>
<td>[13, 14]</td>
<td>[1, 7]</td>
<td>[19, 24]</td>
</tr>
<tr>
<td></td>
<td>[18, 24]</td>
<td>[18, 22]</td>
<td>[11, 12]</td>
<td></td>
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<td></td>
<td></td>
<td></td>
<td>[15, 18]</td>
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</table>

Table 1: Classification of the records shown in Fig. 1.

During the sea trial the environmental conditions where characterized by a wind speed equivalent to a sea state 3 (SS3) and low shipping traffic in the surrounding area. The water depth decreases constantly from 1440 to 230 m through the path. The propulsion mode remained the same for all the records in the selection, just after the 18th it has been switched on the remaining auxiliary machineries, increasing the number of active noise sources in the following records. As one notes the selected noise section is representative for a various set of ship’s navigation conditions. There are both records involved in ship turning at fixed speed and records accounting for ship speed variation with steady heading. Moreover, there are either records acquired during stable navigation condition or records in which occur a simultaneously changing in both the quantities.

While in the first two groups of records a NSL reflecting the corresponding parameter variation is expected, in the third group no relative dependences should be noticeable. Finally, for the records of the last group it is reasonable to predict a superimposed effect on noise level by different contributions. Under such a circumstance it is of interest to see how the noise level is sensitive to this stressing conditions. A summary of the records classification in the above mentioned groups is reported in Table 1.

2.1. Noise spectrum analysis

The noise spectra obtained by four records among those in Fig. 1 are plotted in Fig. 2. Each record falls in one of the groups in Table 1, covering the whole classification. The observed NSL magnitude is greater enough than the estimated level of the ambient noise [11] to assume a negligible contribution of the environmental sources, also at the lower
Fig. 2: NSL in octave bands of noise records corresponding to different ship propulsion and navigation conditions as described in Table 1.

Fig. 3: Self-noise level in the low, medium and high frequency bands for the records in Fig. 1.

Record 16 in Fig. 1 corresponds to the lower ship speed in absence of turnings, same conditions are valid for record 5 but with speed higher of about 5 kn. This speed variation produces a uniform increase of the noise level of about 5 dB, within a range between 3 and 6 dB, for frequencies over 100 Hz. This rise is probably due to the combined effect of a intensification of flow and propeller noise. Further, one observe a growth of noise level at
the lower frequencies also, where the magnitude of propeller blades tonals is excited by an increase of the propeller rotational speed, holding the same pitch.

In Fig. 1 the ship speed at record 9 was the same of record 5, so it allows to compare the two corresponding NSLs and to observe the effect of a ship turning. One notes that during the ship turning the radiated noise level is influenced for the frequencies below 20 Hz. This noise level increase, ranging from 3 dB to 4 dB, may be referred to an increase of the blade tip cavitation due to a non-uniform inflow field surroundings the propellers during the ship manoeuvre.

Finally, it is interesting to evaluate the modification in the radiated noise spectrum when the ship is turning during an acceleration. This specific conditions are met in Fig. 1 by the record 21, which also falls in the interval where all the auxiliary machinery were active. By comparing in Fig. 2 the NSL of record 21 with the others, it is clear that the noise spectrum is subjected to a number of effects that act simultaneously. Observing the four spectra it is possible to distinguish some of these. At the lower frequencies the noise level increases, reaching a major peak. This may be due to a well-developed cavitation at the tips of propellers blades, induced by both the acceleration of blades rate and variations in the inflow regions. At higher frequencies, from 20 Hz to 200 Hz, a rise of the noise level, which probably cannot be related to ship accelerations and manoeuvring, appears. A possible explanation of this variation may be the activation of all the auxiliary machineries, whose rotational and reciprocating mechanisms produces spectral lines and harmonics having a major impact inside this frequency band. The effect of a full propeller cavitation is also evident for frequencies over 100 Hz, where, combined with the flow noise, produces the well-known broadband contribution for the ship radiated noise.

### 2.2. Sonar self-noise analysis

The sonar self-noise has been evaluated in three frequency bands for each record in the sea trial. For each band the magnitude level in dB relative to the mean value for the sea trial is plotted in Fig. 3. Looking at the self-noise in the MF and HF bands, a very similar behaviour is observed, with an maximum variations of ±4 dB. Instead, the self-noise in the LF band shows a profile characterized by a set of distinct peaks that are not visible in the other bands. A more detailed analysis of the self-noise can be carried out by comparing the level variations with the corresponding changing in the ship’s navigation conditions, shown in Fig. 1.

Pointing the attention on the HF band, one notes how the self-noise level follows the variations of the ship speed, even the weakest in magnitude, as for the records from 6 to 10. The self-noise sensitivity in this frequency band is about 1.2 dB/kn and is visible both during periods of ship acceleration (i.e., records from 18 to 19), and deceleration (i.e., records from 13 to 14).

By analysing the self-noise level in the LF band it results less sensible to ship’s speed variations, even if a limited impact still remains especially for high speed values. Instead, a main contribution to the level variation at low frequencies seems to be brought by the ship turnings. A clear case is represented by the peak in the interval between the records 8 and 10, where in absence of speed variations the self-noise increase of about 4 dB with respect to the level in the adjacent records. The same cause-effect phenomena can be observed also in the records from 18 to 22, where another distinct peak is present only in the LF band.
3. CONCLUSIONS

Estimation and analysis of underwater radiated noise represents a very important task in the ship design and later during sea trials. Usually the measurement of ship noise is performed at a certain distance from the source and suffers the estimation of propagation loss via simulation techniques. The chance to measure the underwater acoustic noise radiated by a ship close to its bow represents a great chance to better understand the noise spectral characteristics and to relate them with the actual ship conditions (i.e. propulsion mode and navigation parameters). In this paper it has been presented an analysis of the noise radiated by a ship as recorded at the ship bow. The ship noise has been evaluated in terms of spectral shape, allowing to discriminate the contributions of different acoustic sources or phenomena. Moreover, it has been possible to compute the sonar self-noise level in a selection of frequency bands and to compare this level with the changes in the ship’s navigation conditions. This analysis has allowed to observe a strict dependence of the noise level on the ship speed, for frequencies above 100 Hz, and a distinct increase of the noise level during ship turnings, for frequencies below 100 Hz. Nevertheless, in the observed noise measurements still remain a number of unsolved spectral patterns that appears to be not related with the observed ship parameters. These occasional variations, clearly visible in the sonar self-noise level at low frequencies, may be caused by transient noise phenomena that can affect the noise characteristics under specific circumstances.

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MEASUREMENT SYSTEM TO ASSESS UNDERWATER NOISE FROM VESSELS AND MARINE ACTIVITIES

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Abstract: Underwater noise produced by vessels and platforms can impact mission goals and marine life. For example, excessive noise produced by fisheries research vessels may alert fish and other aquatic animals to the ship’s presence, thereby affecting stock assessments and research objectives. Similarly, vessels relying on underwater acoustic communications systems, sonar operations, and other forms of acoustic exploration require noise from the vessel to not interfere with operations. Vessels and platforms operating in environmentally sensitive areas may also require lower levels of underwater noise to meet desired or imposed criteria; requirements for limiting underwater noise radiation are being applied to new vessel and platform constructions by owners. Regulatory bodies such as the International Maritime Organization and others are also considering regulations on underwater noise for shipping, at a minimum, to combat the rise in noise levels in the world’s oceans.

A highly portable underwater noise measurement system has been developed to assist shipyards, vessel designers, and owners to assess the noise that is produced by new and existing builds. The system can be used to determine compliance with vessel specifications and/or regulations and to identify causes of noise components within the radiated noise spectrum. Measurements with this system comply with the latest regulatory measurement techniques such as ANSI/ASA S12.64-2009 as well as other standards. The system can be deployed from the test vessel using a crane or A-frame, thus eliminating the need for additional vessel support. Measurements can be performed at any convenient location, with the main limiting factor being environmental conditions (e.g. wave height and water depth).

This paper presents an overview of the system design and capabilities, as well as example measurements from testing of a research vessel with a low underwater noise signature. Comparisons of measurements made with the portable measurement system to
measurements from a permanently moored test array are provided, demonstrating the accuracy of the approach.

**Keywords:** Underwater Noise, Vessel Noise, Radiated Noise, Underwater Noise Measurement
INTRODUCTION

Performing an underwater noise measurement that can be used to identify the ‘source level’ or characteristic sound produced by a ship or offshore platform typically requires the collection of at least two pieces of information: 1) sound pressure level data vs. time measured at one or more points in the water and 2) the distance between the source and the measurement point. In most cases, after the measurement is performed, the recorded level is mathematically propagated back towards the vessel to normalize the measurement; thereby requiring knowledge of the distance for such propagation (see References [1, 2]). This allows for assessments of radiated noise compared to a given criteria, diagnostic appraisal, and prediction of noise in unique acoustical environments.

Performing such measurements can be difficult for many reasons. For example, unlike typical measurements of airborne noise, the data collection point is required to be far away from the person collecting the data. This means that hydrophones used to collect sound pressure data must be connected to a cable or other support and positioned somewhere in the water column. This positioning makes the system subject to drag forces, wave induced motion, and other loads that may be unfamiliar to someone who routinely performs airborne noise measurements. Knowledge of the hydrophone location within a tolerance that produces acceptable measurement accuracy must be obtained. Furthermore, tracking of the vessel or platform location as a function of time must be performed if noise measurements are to be collected for an ‘underway’ condition.

Performing underwater noise measurements requires proper setup and planning of the test in order for accurate data to be collected safely. Depending on the complexity of the test and deployment, this planning can take weeks or even months. Factors such as ocean currents, sea state, deployment and retrieval procedures, vessel maneuvering capabilities, data collection and storage requirements, and other factors must be considered. A proper measurement methodology is needed to allow not only for sound data to be acquired but also to continuously acquire data to calculate the distance between the source and receiver. Only with proper planning can an accurate measurement be performed, though unknown factors of ocean currents, wind and weather, and other issues will always present complications.

There are various options available for performing underwater radiated noise measurements of vessels and offshore platforms. All methodologies involve various trade-offs between cost, complexity, and accuracy. A detailed review of available options is presented in Reference [3]; a summary is provided here:

1.1 Permanent Sites with Moored Hydrophones or Arrays

There are multiple facilities throughout the world that are equipped to measure the noise of vessels and some platform types with permanently moored hydrophones and hydrophone arrays. An example schematic of a permanently moored hydrophone array is shown in Fig. 1. One or more mooring weights are secured on the sea floor while a structural cable mechanically attaches the weights to a subsurface buoy. Hydrophones are attached along this vertically suspended cable. A data cable is used to send the hydrophone signals to the data collection point, which is typically located on shore.
The location of the vessel can be determined using a pre-installed GPS device, an Automatic Identification System (AIS), an acoustic pinger system, or other methods. In most cases some connection to the location device is required, whether it be time stamped logging and subsequent retrieval, a radio link, or other method.

The locations of the hydrophones are fixed, more or less, given the fact that the hydrophones are moored in place. Some motion is still possible due to currents; the extent of the motion will depend on the tension in the structural support line, magnitude of the currents, and other factors. In most cases the system has been designed to minimize motions. Some ranges have methods of monitoring hydrophone motion and correcting for it, if determined to be necessary.

The distance between the vessel and each hydrophone (as a function of time) can be directly calculated by combining the position information for both the vessel and hydrophone.

Permanent ranges can be located in regions where nearly optimal acoustic conditions exist. In many cases the water is protected, so third-party vessel traffic can be minimized. Although these locations are not immune to weather events, their influence can be minimized in some cases. These factors help to keep background noise levels low. In addition, water depth does not change (other than tidal changes) which can allow for the characterization of sea floor influences when necessary [4]. As a result of these factors, the data quality is typically excellent when measured at a permanent facility.

Within the USA, the US Navy owns and operates several facilities with permanently moored hydrophone arrays. While these facilities are primarily for military use, they are also available for measurements of commercial vessels. These measurements are often considered to be the ‘gold standard’ of underwater noise signature identification. Other countries also operate such sites.
The primary issues with these facilities from a commercial perspective are 1) they are usually expensive and 2) they can be located remotely and at a great distance from where the vessel is normally operated (or its home port). High costs result from the large amount of setup that is required to install such a facility, as well as other ongoing factors including operating infrastructure, maintenance, number of personnel needed to staff the facilities, and required support vessels. There are additional costs that must be borne by the vessel being tested that include, at a minimum, time and fuel expenses for transit to one of these locations (which may or may not be convenient given a particular vessel’s location) and costs associated with scheduling (particularly for an existing vessel). It is often the case that vessels must travel for days to reach these facilities.

There are other issues to also consider when using these facilities. In many cases time on the range must be booked many months in advance. For military ranges, military projects will take precedence, potentially causing a re-scheduling of the test time. It is often the case that the results of the test are not immediate, and may require months to complete. Lastly, if deficiencies in the noise signature are identified, the test operators on land must communicate with personnel on the vessel being tested in order to perform diagnostics – this can potentially complicate diagnostics if the measured spectrum cannot be seen by all pertinent parties.

1.2 Use of a Support Vessel

An alternative to using a permanent range is to perform measurements from a second, ‘support vessel’. Deployment of hydrophones from support vessels can vary in complexity. A few examples of possible setups include:

- Single or multiple cabled hydrophones suspended at the surface by the support vessel. Vessel can be free floating or anchored. This is a common setup for an ANSI/ASA S12.64 Grade C measurement [1].
- Temporarily moored hydrophone or hydrophone array on support line with anchor at bottom and subsurface buoy at top, cabled to the support vessel. An example is seen in Fig. 2. This could be used for an ANSI/ASA S12.64 Grade A or B measurement [1].
- Single hydrophone mounted on sea floor, as required by DNV Silent Class [2].

For all of these approaches, measurement of the test vessel’s position can be performed using multiple methods, including AIS and/or GPS on the test vessel (as was described for permanent moored systems), radar systems on the support vessel, or even a laser range finder.

The major differences in these approaches are in how the hydrophones are deployed, and the associated accuracy, complexity, and cost. Each of these methods has its advantages and disadvantages.

A single suspended hydrophone is by far the simplest approach, but also has the most drawbacks from an accuracy perspective. Water currents will create a load on the hydrophone and cable. If the vessel is anchored, the hydrophone will move so it is deployed at an angle away from vertical. Even in low currents (less than 0.5 knots) this movement can be significant. Current induced displacement of the hydrophones can be
countered somewhat by adding mass near the hydrophone, though there is a practical limit to this approach. At some point a supplemental line is needed to support the weight, and possibly additional hardware at the support vessel end would also be needed.

![Fig.2: Example schematic of bottom moored array with support vessel](image)

Deployment of multiple hydrophones suspended and supported at the surface only amplifies this problem. More hydrophones means greater drag forces and greater displacements, which would in turn require larger masses to keep the line close to vertical. With these added loads the requirements for support equipment at the surface grows. Such deployments must be considered carefully as damage to the test equipment is possible.

Forces due to currents must also be considered for a free-floating support vessel. It must be recognized that the vessel will be subject to both current and wind loads, and these forces may not occur in the same direction. When the vessel is pushed in one direction and the hydrophone is pushed in a different direction, it can become very difficult to know where the hydrophones are actually located. Again, requirements for support equipment grow in these cases, and damage of equipment can result.

Another factor to consider is wave motion. When hydrophones are suspended from a support vessel they are subject to the vertical motions of the support vessel. Typical, commercially available hydrophones are sensitive to this motion. Support vessel induced hydrophone motion can cause ‘false noise’ in the low frequency region of the spectrum, effectively leading to increased background noise (an example is presented later in this paper). This effect is often amplified for vessels as the vertical motion is non-sinusoidal, resulting in a ‘bleeding’ of false noise over a wider spectral range.

A deployment using a temporarily moored hydrophone or hydrophone array, such as the case shown in Fig. 2, will tend to reduce or eliminate the effects caused by vessel motion in waves. This approach can therefor improve the low frequency range of the measurement by reducing the apparent background noise. This fact is also part of why facilities with permanent arrays have good background noise performance and data quality. However, for a temporary deployment, what this approach gains in accuracy it loses in complexity.
At a minimum, a temporary moored array or single hydrophone would require the support vessel to also be anchored in place. In order to prevent undesirable motions of the vessel in changing wind and currents, a 3-point anchoring system is often required. This may require the use of a third vessel to help set the anchors of the support vessel, increasing cost and time for setup.

In addition, a moored hydrophone is not immune to current effects, and the complete system needs to be carefully designed to ensure the system stays in place for a given current load, and the hydrophones do not have appreciable motion from their assumed locations. This requires a careful balancing act between anchor and subsurface buoy selection, and generally implies very large components. The data cable and other lines connected to the hydrophones must have sufficient slack to allow for vessel movement not only during testing but also during deployment and retrieval. Detailed knowledge of the water depth, and possibly the sea floor composition, is needed prior to deployment. Proper design of this setup is not trivial, and complications due to inaccurate environmental information should be expected.

Whenever a support vessel is used, noise from that vessel must be minimized. This not only means securing all equipment and running on battery power (or possibly a small, isolated generator) but also means somehow minimizing other noise such as waves slapping against the side of the hull. This may not be possible in all situations.

In summary, when using a support vessel, deploying and supporting hydrophones from above is a simpler method but can lead to inaccuracies in the measurement due to current, wind, and wave loads and motions. These effects become amplified quickly in areas/days with degraded weather conditions. Bottom mooring the hydrophones reduces or removes these effects, but the complexity of deployment is greatly amplified. These factors lead to increased net cost of the measurement, and potentially a loss in accuracy.

**A NEW APPROACH**

A new approach to underwater noise measurements has been devised which attempts to capture the cost benefits of performing measurements from a support vessel (vs. a permanent range), the simplicity of a surface supported hydrophone deployment, and the accuracy of a moored array, while maintaining or improving portability. The authors have developed a new measurement platform type, the Buoy Acoustic Measurement System (BAMS). This system uses an independent, floating buoy to support single or multiple measurement hydrophones as well as positioning and data acquisition electronics. Details of the design and use are provided in the following sections.

**2.1 Overall Design**

Pictures of the BAMS platform are provided in Figs. 3 & 4, and a schematic of a typical deployment is shown in Fig. 5. In this particular example the system is freely floating (not moored), and would comply with an ANSI/ASA S12.64 style measurement. Moored configurations are also possible, but are not the focus of this paper.
The system has the capability to collect data from multiple hydrophones suspended from the surface buoy. This data is processed and stored on-board, and is simultaneously transmitted via radio link to the BAMS operators located on the test vessel. The collected data from a particular test run can be analyzed immediately, and indications of acceptance to a particular criteria or diagnostic information can be determined within minutes of performing a run. Since the BAMS operators are located on the test vessel, items identified as being possible issues can be immediately inspected. This is particularly useful when identification of problem ship or platform components is required to be performed in an efficient manner.
The floating platform design simplifies setup, deployment, and retrieval as compared to other deployment methods. First and foremost, BAMS completely removes the need for a support vessel. BAMS is designed to be deployed and retrieved from the vessel being tested using a crane, A-frame, or other similar lifting device. The system can be easily lifted by two people, so the lifting capability of the vessel’s equipment does not need to be extreme.

The BAMS design has a minimal area above the water surface so wind loading is small relative to current loads. This drastically reduces the likelihood of different wind and current loading moving the hydrophones to unknown locations, thereby improving positioning accuracy relative to similar methods using a support vessel.

Currents are still a factor with this design, though primarily during deployment and retrieval. Once BAMS has been deployed, the hydrophone array will stay directly below the surface buoy when the current profile vs. depth is roughly uniform. This can be understood by imagining the following situation. If the system is held still in a strong current, there will be varying forces on the different components in accordance with their coefficient of drag and cross sectional area. This will cause these components to accelerate in the direction of the current at different rates. However, once each component reaches the same velocity as the current, this force will drop to nearly zero. Therefore, at equilibrium, all components will be moving at the same velocity, and gravity will ensure that the hydrophones are beneath the surface buoy. The hydrophone positions relative to the buoy are known based on the hydrophone cable length.

In typical measurement locations, currents range from 0-2 knots, meaning BAMS will also be traveling at this speed. As described in both ANSI/ASA S12.64 [1] and DNV Silent Class [2], the test vessel is required to pass in a straight course past the
measurement hydrophones. Based on experience using this system, it is relatively easy to maneuver a vessel through a course and obtain a target “Closest Point of Approach” (CPA) of 50-250 meters or more (depending on the needs of the test) even when the buoy is moving at 2 knots. For measurement of stationary items such as offshore platforms operating in a drilling or production mode, the movement of the buoy can be beneficial as this facilitates multiple measurement aspects and distances.

BAMS is designed to be portable, but strong enough to withstand the forces imposed during deployment, operation, and retrieval. BAMS can be completely broken down and stored in multiple shipping cases. The design goal for each case is to allow for air shipment using commercial shippers, and to make them light enough to be transported by no more than 2 engineers.

2.2 Buoy Design

The BAMS buoy component is designed as a limited buoyancy wave motion damper. This results in waves washing over the buoy and reduces the vertical motion of the system in response to normal wave motion. By reducing the vertical motion of the buoy, the vertical motion of the hydrophones and increase in low frequency background noise previously discussed are also reduced. This results in better data quality as compared to measurements performed from support vessels (non-moored deployments).

An example is shown in Fig. 6. Background noise measurements performed on different days, varying sea states, using different test configurations are shown here. The BAMS and “dive boat” measurements were both performed in seas roughly corresponding to Sea State 4 (approximately 1.5 meter wave heights). It is seen that the dive boat measurement suffers far greater ‘false noise’ due to wave motion as compared to the BAMS system, with differences ranging up to 20 dB. Comparisons of BAMS to a measurement performed from a small fishing boat in Sea State 2 (less than 0.5 meter wave height) shows roughly 5-20 dB improvements at low frequencies. (Note that the differences in levels above roughly 100 Hz are due to actual background sounds. For example, the noise measured from the fishing boat contained sounds from snapping shrimp.)

This data shows the BAMS design can provide significant reduction in the effective background noise and an improvement in measurement accuracy at low frequencies as compared to other approaches where the hydrophone is supported at the surface. However, there is clearly some component of ‘false noise’ that results from the residual motion of the hydrophone even for the BAMS system. (Unfortunately, the actual background noise at low frequencies was not obtained in these examples.) This false noise may cause problems for some measurements, particularly when measuring vessels with very low radiated noise or measurements performed at large distances from the test vessel. A moored hydrophone approach will provide lower background noise than a free-floating BAMS. However, given the simplicity of BAMS deployment and use, the extra effort required for designing, deploying, and carrying out a test using a moored hydrophone deployment must be worth the extra cost and effort. In most cases the free-floating BAMS approach is more than adequate, particularly in lower sea state conditions.
The buoy platform has a rigid aluminum skeleton that was designed to take the extreme forces associated with launch and recovery in heavy seas. It contains built-in attachment points in the event that a test setup is used where the hydrophones are moored in place. This allows for flexibility when testing to standards that require this style setup, or for testing in conditions where a secured hydrophone location is advantageous.

2.3 Electronics

The BAMS platform supports multiple electronic equipment items for data acquisition, positioning, and communication. These items include:

- Low noise power supply for hydrophones
- Multichannel signal acquisition, conditioning, & processing equipment
- GPS telemetry system
- Telecommunication transceiver to secure a communication link between BAMS and the vessel being tested
- A VHF-based transponder to send the position of BAMS to the chart plotter of the vessel being tested
- Computer with external hard drive
- Redundant battery systems for power and battery monitoring equipment

The data acquisition system is used for collecting and processing the signals from the hydrophones. The position of the buoy is constantly calculated using GPS receivers and
monitoring software; this, combined with the known hydrophone depth(s) allows for accurate accounting of the hydrophone position(s).

A second, VHF based positioning system is used to communicate directly with the vessel, allowing for the BAMS location to be monitored using standard chart plotters. This assists the vessel’s captain in plotting a course to attain the desired CPA for a particular run. This information can be augmented by the vessel’s radar; BAMS contains a radar reflector as a backup for vessel navigation.

The telecommunications transceiver is used to establish a data link with BAMS operators located on the test vessel. The test vessel is also outfitted with a telemetry system and telecommunication transceiver so the BAMS operators can monitor the vessel location and communicate with the buoy to identify when to start and stop collection of data. Knowledge of both vessel and hydrophone locations allows for calculation of distance for source level computations. The communications link with BAMS can be maintained over several miles to allow for good background noise measurements.

All data acquisition, positioning, and telecommunications operations performed by BAMS are processed, controlled, and coordinated through an on-board computer. The computer runs custom software that has been created specifically for BAMS operation. The computer also allows for rapid processing of information and facilitates test initiation, test completion, and data storage.

The BAMS electronics are housed in watertight cases that are mechanically attached to the buoy. The electronics cases have been tested to a submergence depth of fifteen meters for twelve hours. Waterproof connectors penetrate the electronics cases and allow communication between different pieces of equipment.

**CASE STUDY**

Recently, the authors had an opportunity to compare a measurement of a vessel using BAMS to a measurement of the same vessel performed at a facility using a permanent hydrophone array. The measurements were performed within a few months of each other. This vessel is a research vessel with a low underwater noise signature as compared to common commercial vessels such as tug boats, tankers, platform supply vessels, etc.

A comparison of the measurements is provided in Fig. 7. Third-octave band measurements are shown here. Both measurements were performed using a 3-hydrophone array using the basic test framework outlined in ANSI/ASA S12.64 [2]. The primary difference in measurement methodology is the BAMS measurement used a nominal CPA of 100 meters where the permanent array facility used a nominal CPA closer to 150 meters. The spectra that are shown have been distance corrected using the procedure outlined in ANSI/ASA S12.64.

Although the absolute magnitude of the spectrum cannot be shown, the scale is seen to be 5 dB per division. In general the correlation between measurements is nearly exact. Small deviations on the order of 1 dB are seen at specific frequencies between 25 and 20,000 Hz. Unfortunately, there were some differences in the operating machinery between measurements. Further analysis indicates that this is the reason for the
discrepancy at 20 Hz. Differences at 10 and 12.5 Hz are due to hydrophone motion in the BAMS measurement.

![Comparison of Permanent Range and NCE BAMS UWRN Data](image)

**Fig. 7: Comparison of distance corrected measurement, BAMS vs. permanently moored hydrophone array facility**

This comparison shows that the BAMS system can provide accuracy on the same level as facilities with permanently moored hydrophones. The data shows that both the collected hydrophone data and relative positioning information between the hydrophones and the test vessel are accurate for BAMS.

It is worth noting that the cost of performing the BAMS measurement in this example was significantly less expensive than performing the measurement at the permanent hydrophone array facility. This estimate includes not only the costs of the facility rental but also costs associated with the multi-day transit from the vessel’s home port to the facility.

Furthermore, the BAMS measurement was able to be performed while the test vessel was transiting between two locations as part of its routine schedule. Testing itself took one day to perform; setup and teardown of the BAMS system was performed in a manner that did not impact the vessel’s normal schedule.

**SUMMARY**

A new method of performing underwater noise measurements of vessels and offshore platforms has been introduced. The system allows for measurements to be performed in accordance with ANSI/ASA S12.64 as well as other measurement standards. The measurement system, called BAMS, is an improvement on existing approaches in the following ways:
• Accurate measurements, shown to be nearly identical to a facility using a permanent hydrophone array
• No support vessel is needed
• The system is portable and can be deployed using a small crane, A-frame, or similar lifting device
• Knowledge of hydrophone position is inherent in the design
• All data acquisition systems are contained within the measurement buoy
• Collected data can be stored on-board the buoy and transmitted to the test vessel for immediate analysis
• Initial results of a test run can be obtained within minutes of completing the test run
• The location of the buoy (and measurement hydrophones) can be identified on most vessel’s chart plotters using standard equipment, allowing for real-time course correction if needed.
• Cost of measurement is a fraction of other methods
• Measurements can be performed in a wide range of locations, and can accommodate vessel schedules.

The primary objectives of the BAMS design have been achieved, though some work is still necessary to further improve the performance at low frequencies. Although BAMS provides an improvement to other methodologies where hydrophones are supported at the surface, it is not immune to the effects of wave motion. This causes ‘false noise’ at low frequencies, effectively raising the background noise at the test site.

Formally this is a limitation of BAMS; in poor weather conditions BAMS measurements may be limited at low frequencies. However, it is common for vessels with underwater noise criteria to have requirements indicating measurements should be performed in calmer waters. In addition, vessels with higher levels of noise such as container ships, tankers, supply vessels, high speed craft, etc. may not suffer from these effects as the vessel noise would likely be higher than the background, false or not. Lastly, since the levels of false noise are dependent on sea state, measurements in calm waters are of the same caliber as moored hydrophone arrays.

This system will allow owners, operators, and other interested parties to determine the underwater signature of any vessel or platform in an accurate and economic manner. This signature may be important with respect to increased interest in ships operating in environmentally sensitive areas or with respect to future regulations on underwater noise being considered by governments and IMO.

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INTEGRATED TOOL FOR THE ACOUSTIC ASSESSMENT AND MONITORING OF MARINE ACTIVITIES AND OPERATIONS

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Abstract: A combined tool to predict and monitor the sound produced by offshore facilities is being developed. The aim is to provide a top quality, integrated solution to pre-assess, monitor and mitigate the acoustic impact of offshore energy parks, or other marine operations, while allowing an optimization of costs. The system consists of a monitoring tool (buoy), which is able to autonomously collect and send in-situ acoustic data to shore, and a numerical simulation tool, whose results can be used during the design phase of the facility to predict the acoustic impact and during the operation phase to complement the information collected by the hydrophones. The monitoring tool comprises a buoy, or an array of buoys, equipped with hydrophones and other environmental sensors, self-sufficient power and the data recording and transmission components. The buoy design focuses on maximizing the autonomy of the tool, improving the reliability of the data stream and generating the information needed to optimize the frequency and success of maintenance operations therefore reducing the overall cost. The acoustic data can also be used to detect anomalies in the mechanical components of the energy devices, by identifying deviations in their acoustic signature. The simulation tool computes 2D and 3D sound maps produced by multiple noise point sources, for medium-sized sea domains. The underwater propagation phenomenon is modeled through the Helmholtz equation, which considers the wave physics, and includes the seawater absorption. Non-uniform sound speed vertical profiles and adjustable bottom reflection properties are also included in the model, making it more realistic than other state-of-the-art alternatives. The numerical method includes a set of plane waves working as base functions, providing a superior accuracy with an affordable computational cost. For optimal
results the simulation tool will be calibrated in-situ through acoustic data provided by the monitoring tool.

Keywords: acoustic monitoring, acoustic impact assessment, long-term monitoring, real-time monitoring, sound propagation modelling, partition of the unity method

1. INTRODUCTION

In order to reduce greenhouse gas emissions and to achieve the goals of the Kyoto Protocol there has been a global effort to invest in new and less polluting sources of energy. This has been reflected in the implementation of legislation to promote renewable energy, namely the Renewable Energy Directive 2009/28/EC in Europe and the 2005 Energy Policy Act in the USA. A part of this investment corresponds to research and implementation of offshore marine renewable energy [1] as it benefits from the constant resource availability and low spatial constraints when compared with in-land harnessing.

Strong winds regularly blow over the open ocean making one of the current goals of offshore wind industry to locate wind farms as offshore as possible which brings operational and monitoring constraints, as well as concerns about the potential impacts on the marine environment. One factor of concern is related with the potential contribution of man-made noise to the underwater acoustic environment. From a biological point of view this is particularly relevant since most marine animals rely on it to survive [1]. This concern is reflected in several international legislations and agreements. Two examples are the Marine Strategy Framework Directive (MSFD) and the OSPAR Convention. Both address the need to continuously monitoring low frequency sounds in 1/3 octave bands, mainly those centred at 63 Hz and 125 Hz. Noise mapping based on field measurements and modelling [2] is also recommended.

Considering that the noise emitted by devices is dependent on sea state continuous data acquisition, it is crucial to understand the link between natural background noise and anthropogenic noise. Furthermore, permanent acoustic monitoring of an energy park offers the opportunity to use this data as a diagnostic tool of the park itself. Most mechanical and structural issues that may occur within a device are expected to have an acoustic output. This means that a noise will be superimposed on the devices normal acoustic signature. Identifying and flagging these noises can be a useful remote diagnostic tool to detect malfunctions in the energy park. In addition, if more than one monitoring device is present, the results can be triangulated to detect which device needs intervention.

The environmental tool, presented in this paper, is being developed to allow developers to predict, collect and analyze environmental parameters in an integrated and consistent manner to assess the “health” of the park, or other marine activities requiring an acoustic assessment, and acoustic impacts on the marine environment. For this the tool comprises two components: one of real-time data acquisition; the hardware component, and other of acoustic simulation; the simulation component.

The hardware component includes a set of sensors for real-time data acquisition which are supplied by an autonomous power system. Currently, this comprises a hydrophone able to record and perform the acoustic data analysis in real-time. The other sensors were selected to complete to complement the acoustic information as well as to provide information to operations and maintenance. However, the selected sensors may be modified to fit different needs.

The simulation tool is being designed to satisfy two requirements of the integrated product. On the one hand, during the design stage, it predicts the acoustic impact from preliminary data of the noise sources. This prediction will be a valuable input when
determining the farm location and layout configuration in order to prevent conflicts with other economical activities or to protect specific biological resources. On the other hand, during the operational stage, the simulation tool can also compute the underwater noise field from experimental data recorded by the monitoring tool at several spots. This will provide an estimation of the spatial acoustic impact of the marine operation.

This paper presents a brief description of both tools and the advantages that they provide when deployed together. Regarding the monitoring tool, focus is placed on the acoustic components and how they fulfil market needs. Regarding the simulation tool the basis of the model and its numerical implementation are summarized.

2. THE HARDWARE COMPONENT

The hardware component of the acoustic monitoring tool comprises of an autonomous buoy equipped with a set of above and below water sensors for real time data acquisition and a telemetry system for data retrieval (see Fig. 1). These are supported by a power system and a data processing unit. Other components of the tool are a UHF onshore station, acting as a gateway, to forward the data received from the buoy onto an on-line cloud platform which stores all the data and performs data interpretation and reporting (web page, email, text message). This interface also allows an integrated system management by permitting the whole system to be configured remotely. A terminal for onsite access for maintenance and system management (e.g. from a boat) will also be provided (see Fig. 2).

![Monitoring tool design: view above water (left) and view underwater (right).](image)

The buoy is an adapted floating navigation aid model which has been robustly proven offshore with a foam filled hull making it unsinkable in case of breach. The central load bearing structure is made from galvanized steel and provides an intrinsically stable configuration with an integrated counter-weight. It includes a self-contained lantern and radar reflector to comply with IALA recommendations.

The DC voltage that powers the electrical equipment of the buoy is supplied by a power system that includes a photovoltaic panel array, a solar charge controller; that regulates the generated voltage and current, and a battery bank that stores the energy and powers the electrical equipment. Protection devices are installed to prevent over current. The total peak power installed is 280 watts and the capacity of 400 Ah ensures autonomy of over 15 days in cloudy conditions.

The telemetry system is based on a UHF radio link that can work up to 20 km offshore, and includes the possibility of GSM and Inmarsat data communications to accommodate
different locations and requirements. Both GSM and Inmarsat connect directly to the public network. A Globalstar satellite tracking device is used for geofencing and ZigBee communications for local maintenance.

![Fig. 2: The monitoring tool.](image)

The buoy data system architecture is presented in Fig. 3 and consists of a CAN bus that interconnects the main devices in the buoy (sensors, data logger and power system). This topology uses a gateway that converts the specific sensor interface (e.g. RS232, analogic) and protocol (e.g. NMEA) into CAN a normalized protocol, allowing for easy replacement of sensors or introduction of new one. The data logger is the main processor unit that drives the bus communications and telecommunications, manages and stores data and programs on-board non-volatile storage. The telemetry system devices connect directly to the data logger by RS232, except for the Globalstar device that works autonomously.

The above water instrumentation includes a weather station, a solar radiation sensor and an accelerometer. A water temperature sensor, a transmissometer and an ADCM (acoustic Doppler current meter) as well as a hydrophone the in-water comprise the below water instrumentation, see Table 1.

![Fig. 3: The data system architecture.](image)
The hydrophone was specifically developed for this project and is able to record underwater acoustic signals in the frequency band from 1 Hz to 49.8 kHz. This frequency range is suitable to record the underwater noise emitted by different anthropogenic activities, as well as by some biological signals.

Both resolution and PGA gain can be adjusted which allow the hydrophone to have maximum sensitivity without data saturation. Based on the long-term data assessment objective the hydrophone was designed to record and analyse real-time acoustic data both in terms of 1/3 octave bands and narrow band. This is of particular interest since narrow band analysis is important to identify potential tonal noise with origin on devices and 1/3 octave band analysis serve the purpose of environmental impact assessment. Furthermore, this feature allows low size data output. As an example, a 5 minutes sample that would generate a 60 Mb wav file, is reduced to 110 Kb for an averaging time of 0.1 s, resulting in a reduction of over 500 times. Depending on the sampling regime it is possible to select the average of the acquired data over different time periods.

Another big advantage of this tool is the possibility to tailor the sensor array and configuration to the markets need. Also the combination of sensors variety together with the versatility and reliability of the buoy, allied with all the possible transmission systems makes it a powerful tool for accessing real-time, site-specific crucial information in underwater acoustic data assessment.

<table>
<thead>
<tr>
<th>Sensor</th>
<th>Parameters</th>
<th>Benefit</th>
</tr>
</thead>
<tbody>
<tr>
<td>Hydrophone</td>
<td>Underwater sound frequencies and SPL</td>
<td>Acoustic impact assessment, device surveillance</td>
</tr>
<tr>
<td>Weather station</td>
<td>Wind speed and direction, air humidity and temperature, dew point, barometric pressure, heading, pitch and roll, geographic position data</td>
<td>Complement for acoustic data, information for O&amp;M</td>
</tr>
<tr>
<td>Accelerometer</td>
<td>Wave height and period data</td>
<td>Complement for acoustic data</td>
</tr>
<tr>
<td>Water temperature</td>
<td>Water temperature data</td>
<td>Complement for acoustic data</td>
</tr>
<tr>
<td>ADCM</td>
<td>Current direction and speed data</td>
<td>Information for O&amp;M</td>
</tr>
<tr>
<td>Transmissometer</td>
<td>Turbidity data</td>
<td>Visibility for underwater operations</td>
</tr>
<tr>
<td>Solar irradiance</td>
<td>Solar irradiance data</td>
<td>Solar production estimation</td>
</tr>
</tbody>
</table>

Table 1: Description of the above and in-water instrumentation/sensors.

3. THE SIMULATION COMPONENT

The simulation tool computes 2D and 3D sound maps produced by multiple noise point sources placed at the surface, for medium-sized sea domains. The underwater propagation phenomenon is modelled through the Helmholtz equation, which considers the wave physics, including interference, reflection and refraction, see details in [3]. Sound must be harmonic for each computation and, for this reason, the input noise is decomposed into several frequency bands of a spectrum, and a computation is executed for each band.

The seawater absorption is estimated using the simplified formula by Ainslie & McColm [4], and introduced into the model through the imaginary part of the wavenumber. The model accepts a non-uniform sound speed vertical profile, depending on the water temperature, salinity and the depth. The sound speed is obtained using the UNESCO equation.

The computational domain is delimited by six boundaries. Adjustable reflection properties are prescribed at the sea surface and sea bottom. Artificial boundary conditions are prescribed.
at the four lateral boundaries. Specifically, the Perfectly Matched Layer (PML) method is placed at each lateral artificial boundary in order to achieve full absorbing behaviour.

The boundary value problem resulting from the mathematical model is solved using the Partition of Unity Method (PUM), a variation of the Finite Element Method, following the procedure present in [5]. The PUM allows the inclusion of specific shape functions, containing \textit{a priori} information of the solution. In the case of the Helmholtz equation, it is advantageous the inclusion of sets of plane waves pasted at each node of the mesh. In the 3D case, the plane waves directions are generated following the algorithm created by Leopardi [6].

The use of plane waves acting as basis functions allows the generation of coarser meshes with several wavelengths per element size and provides a higher accuracy to the tool. Due to the oscillatory nature of the sound propagation problem, special attention is focused on the computations of the integrals involved in the elemental matrices. To speed-up this process and make it affordable a set of semi-analytical integration rules have been developed.

To increase the accuracy of the simulation tool, it will be calibrated in-situ through acoustic and environmental data provided by the monitoring tool. Parameters such as the water temperature and salinity, and the geology of the bottom can be adjusted in order to improve the quality of the results.

To illustrate the capabilities of the simulation tool we present two examples. In the first example (see Fig. 4) we compute the sound pressure level generated by 2 sources producing a pressure of 1Pa. at 10 meters. The depth is 100 m at the centre of the figure. The bottom slope is 0.02 and its transmission coefficient is 0.8. Fig. 5 shows the 3D sound pressure level map of a single source over a partially reflecting bottom. The frequency is 1 kHz at both examples.

![Fig. 4: Sound pressure level field produced by two sources over a sloping bottom.](image-url)
Fig. 5: 3D SPL map produced by a single source with partially reflecting bottom.

4. CONCLUSIONS

The integrated tool under development will be an excellent way to predict and control the acoustic impact of marine activities or operations. The tool takes advantage of the integration of a monitoring tool which is able to record \textit{in situ} measurements, and a simulation tool which is able to predict and generate noise maps in the area of interest. Together they offer a robust, precise and integrated way to carry out underwater acoustic assessments.

5. ACKNOWLEDGEMENTS

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SIRAMIS: PRELIMINARY ANALYSIS OF ACOUSTIC AND SEISMIC SHIP SIGNATURES


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Abstract: SIRAMIS is a project coordinated by the European Defence Agency standing for Signature Response Analysis on Multi Influence Sensors. As most of the international trade is carried out through marine routes, it is important to evaluate the vulnerability of the merchant vessel fleet to sea mines in order to be able to limit the potential exposure to this threat and apply effective mine counter measures like Target Simulation Mode mine sweeping.

In this project, the participating nations pool their measurement and analysis capabilities to improve their knowledge on the underwater signatures of merchant vessels and understanding of the near field ship signature in relevant and realistic scenarios.

The project involves a series of recording campaigns performed near shipping lanes in the national waters of the participants, using various multi-influence measurement systems. The data analysis will help to separate the effect of the differences between the measurement systems and the environments from the features specific to the measured ships. A further analysis will investigate the relationship between the merchant vessels signatures and their characteristics.

This paper presents the initial results of the acoustic and seismic signature analysis.

Keywords: Merchant vessels, acoustic signature, seismic signature.
1. INTRODUCTION

SIRAMIS is a project coordinated by the European Defence Agency standing for Signature Response Analysis on Multi Influence Sensors. As most of the international trade is carried out through marine routes, it is important to evaluate the vulnerability of the merchant vessel fleet to sea mines in order to be able to limit the potential exposure to this threat and apply effective mine counter measures like Target Simulation Mode mine sweeping. In this project, the participating nations pool their measurement and analysis capabilities to improve their knowledge on the underwater signatures of merchant vessels and understanding of the near field ship signature in relevant and realistic scenarios. The project involves a series of recording campaigns performed near shipping lanes in the national waters of the participants, using various multi-influence measurement systems. The data analysis will help to separate the effect of the differences between the measurement systems and the environments from the features specific to the measured ships. A further analysis will investigate the relationship between the merchant vessels signatures and their characteristics. This paper presents an overview of the collected data and preliminary results of the analysis of acoustic and seismic data.

2. OVERVIEW OF COLLECTED DATA

Opportunistic measurements of merchant vessels were performed using measurement systems from the various participating nations deployed in shipping lanes. The location of the measurement campaigns are presented in Fig.1.

The data from Automatic Identification Systems (AIS), whose use is mandatory on large vessels, were also recorded. The AIS data includes information about the vessels, such as the length and gross tonnage and information on its trajectory based on their on-board global position system (GPS) receiver. The latter enables to determine the vessels passing in the vicinity of the sensors, as well as their distance to the hydrophone at the Closest Point of Approach (CPA). These opportunistic measurements enabled the collection of a large amount of acoustic, seismic, hydrostatic, electric, and magnetic data for a variety of types of merchant vessels, representative of the merchant vessel fleet sailing in the European waters: cargo, RoRo, tankers, ferries, fishing boats, tugs...

In addition to the opportunistic measurements, some specific measurements were using controlled ships and controlled sources as well as measurements with the various mobile systems at the same locations (benchmark). These measurements enable comparison of the
measurement systems, of the environments they are deployed in and of the variability from run to run as well as the influence of the distance at CPA.

3. ACOUSTIC SIGNATURE AND ACOUSTIC INDICATORS

In addition to the information related to the vessel being measured, the acoustic signal recorded by a measurement system contains information related to the propagation from the vessel to the sensor and on the ambient noise at the time of measurement. Consequently, the acoustic signal can hardly be considered to be the acoustic signature of the vessel.

Various mechanisms contribute to the acoustic radiation of ships [1], among which propeller cavitation is often dominating. Other contributions are due to machinery noise and flow noise. All these mechanisms lead to the generation of narrow band and broadband components, whose level and directionality varies with the speed and the mode of operation of the vessel, and, of course, from ship to ship. The received signal results from the propagation of these components and varies with the propagation distance, thereby presenting different characteristics in the far field and in the near field. All these aspects relate to the “acoustic signature”, which is not a well-defined concept.

Comparison of the acoustic signature of ships requires the definition of quantitative indicators that enable such comparison. In order to perform such comparison, the analysis team of the SIRAMIS defined a number of indicators based on the narrow band spectrum and One Third Octave (OTO) spectrum. The selected indicators do not cover the full scope of the acoustic signature, but they enable quantitative measurements and comparison. One of these indicators is the Radiated Noise Level (RNL), which is obtained from the temporal maximum of the OTO spectrum after a correction for the measurement distance [2]:

\[
RNL(f) = \max_t OTO(f, t) + 20\log_{10}(r_{CPA}/r_0)
\]  

Where \(OTO(f, t)\) is the OTO spectrum computed as a function of time \(t\) and frequency \(f\) and the other term is the correction for the measurement distance. The latter assumes only spherical spreading loss over a distance \(r_{CPA}\), which is the distance at CPA, and \(r_0\) is the reference distance equal to 1 m.

A similar approach is adopted regarding analysis of the seismic signals, measured by sea floor mounted accelerometers. However, since the amount of scientific data available regarding this topic is limited, the correction for measurement distance, which is subject to debate regarding the acoustic signature [3], is not applied to the seismic analysis.

4. SELECTED RESULTS

4.1 Variability from run to run

Some of the controlled measurements included several runs of the same ship at the same speed over a measurement system. Fig. 2a presents the maximum of the OTO spectrum recorded for four measurements of the same ship, with varying horizontal offset at CPA, all less than 25 m (hydrophone depth is 21 m). This figure illustrates the
repeatability of such a measurement, which is of importance when comparing measurements of different ships. Observed differences between ships are significant only providing they exceed this repeatability. It is seen that the overall shape of the measured spectra is fairly independent of the measurement, with a spread mostly below 5 dB at frequencies above 10 Hz.

4.2 Variability due to the environment

In one measurement campaign, four measurement stations equipped with the same sensors were deployed at separate locations within about 200 m from each other. The water depth at each measurement station is different. Since the same ships were measured at these measurement stations, this enables to gain some insight in the variability of the measurements due to a change in the environment. Note however that larger differences can be expected when the measurement campaigns are performed in truly separate locations, with different sediment properties.

Fig. 2b shows measurements performed with the same ship at the same nominal speed. Each curve corresponds to a different measurement station and is the average of measurements with horizontal CPA distance below 25 m. The received levels were converted into RNL using formula (1). The spread between the curves is mostly below 10 dB at frequencies above 10 Hz, which is larger than the run to run repeatability.

4.3 Effect of speed and propulsion type

Fig. 3 illustrates the influence of ship operation on the RNL. In Fig. 3a, the RNL of a ship at two different speeds is presented. Both curves are obtained as the average of several runs with horizontal CPA distance below 25 m. In this example, the higher speed causes an increased RNL in most frequency bands, sometimes of up to 20 dB.
Fig. 3: Variability due to speed and propulsion type: (left) same ship at different speeds, (right) same ship with different modes of engine operation.

Fig. 3b is also obtained from an average of runs with horizontal CPA distance below 25 m. It corresponds to a ship sailing at 7 knots whose engine can operate in two distinct modes: diesel electrical and diesel mechanical. In the diesel mechanical mode, the torque from the diesel engine is used to rotate the propeller; in the diesel electrical mode, the torque is used to generate electricity, which is used (possibly among other things) to power an electrical motor that operates the propeller. The latter mode is significantly more quiet than the diesel mechanical mode: the difference is between 20 and 35 dB for frequency bands over 10 Hz. The possible cause of this differences is propeller cavitation which is more pronounced in diesel mechanical mode due to the fact that the propeller pitch have to be reduced in order to sail at low speed in this mode.

4.4 Seismic signal / effect of propulsion type

Fig. 4a presents the maximum of the OTO spectrum of the seismic acceleration signal for the same ship as in Fig. 3b. The X-, Y- and Z-components correspond to motion in the track direction, perpendicular to the track direction and in the vertical direction, respectively.
Fig.4: Effect of propulsion type on the seismic signal shown as an OTO spectrum for each component (left) and as a narrow band spectrum for the X-component (right).

They present a similar level within 10 dB. The shape of the OTO spectra is similar to that of the acoustic signal; the difference may be attributed to the frequency response of the seismic sensor. The variation between the two modes of operation is of similar amplitude as for the acoustic signal. Both observations support the idea that the seismic and acoustic signatures are due to the same physical mechanisms on the ship. Fig. 4b presents the narrow band spectrum of the X-component. It indicates that, although the OTO spectra present similar shapes in the two modes of propulsion, the frequency content is dominated by tonals in the diesel electrical mode whereas broadband components dominate in the diesel mechanical mode, due to propeller cavitation.

5. CONCLUSION

The preliminary analysis of the SIRAMIS data demonstrated a run to run variability of the order of 5 dB and a variability due to the environment that is somewhat larger, up to 10 dB. Significant differences in RNL, i.e. much larger than the measurement variability were observed when changing the speed or the mode of operation of ships. The analysis of the seismic data supports the acoustic observations and indicates that both signatures are related to the same physical mechanisms. Further analysis of the data will be devoted to finding grouping and scaling relationships to model the ship signature as a function of ship type and parameters as in [4]-[6].

6. ACKNOWLEDGEMENT

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PREDICTION OF PROPELLER RADIATED NOISE BY ONBOARD MEASUREMENT

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\textbf{Abstract:} Propeller radiated noise from the ship is regarded as major noise source to disturb the ocean ecosystem. In order to preserve the oceanic life, MEPC (Marine Environmental Protection Committee of IMO) has been drafting the underwater noise regulation since 2008. Accordingly, how to predict propeller radiated noise exactly is an important issue. Conventionally, the propeller radiated noise is measured directly using hydrophones located in the sea. This direct method requires much time and cost for the installation and maintenance of the measurement system. Furthermore, it is difficult to measure the propeller radiated noise under the bad weather condition.

This paper proposes an indirect prediction method of propeller radiated noise by onboard measurement. The method can largely save the time and cost compared with the conventional direct method. According to the proposed method, firstly, the sound transmission coefficient of a hull structure is measured in dry dock to estimate the transmitted noise from outside to inside the ship. Secondly, the noise and vibration inside the ship is measured during the sea trial, followed by the ambient noise separation. Finally underwater propeller noise is predicted by using the sound transmission coefficients and measured onboard noise and vibration signals. The feasibility of proposed method was verified through the cavitation tunnel experiments. The method was applied to the prediction of propeller radiated noise during sea trials for two kinds of ships. From the results, it is expected that the proposed method may enable to predict propeller radiated noise with ease, thus to contribute the propeller noise control.

\textbf{Keywords:} Propeller Radiated Noise, Onboard measurement, Noise Estimation
1. INTRODUCTION

Underwater radiated noise from ships is regarded as a major noise source to disturb the ocean ecosystem [1]. In 1991, IFAW (International Fund for Animal Welfare) firstly recognized that underwater radiated noise by ship can harm marine life. Thereupon, they published the report ‘Ocean noise: Turn it down’ which regulates the ship noise as the oceanic pollution. In response to this, MEPC-IMO (Marine Environmental Protection Committee of International Maritime Organization) has been drafting the underwater noise regulation since 2008. In order to preserve the ocean ecosystem and cope with the regulation in future, it is essential to develop the technology to reduce the underwater radiated noise by ships. Prior to this, it should be predicted how much noise from ships is radiated to the sea exactly.

Among various noises radiated from ships, it is said that the main source is the propeller, especially the propeller cavitation [2]. Accordingly, how to exactly measure and predict noise by the propeller cavitation is currently an important issue. Conventional method is using hydrophones located in the sea far from the ship thus to measure the propeller radiated noise during ship’s cruise. This method shows high reliability since it directly measures the noise. The method, however, requires much time and cost for the installation and maintenance of the measurement system. That is, the structure to fix hydrophones should be firstly manufactured and the measurement process requires several equipment such as buoy, test ship, GPS system for hydrophone localization. A lot of human efforts are also required to install heavy hydrophone array. Furthermore, it is difficult to measure the propeller radiated noise under the bad weather condition due to noise signals.

In order to improve the previous direct measurement method, this paper proposes an indirect prediction method of propeller radiated noise by onboard measurement. According to the proposed method, firstly, the sound transmission coefficient of a hull structure is measured in dry dock to estimate how much propeller radiated noise is transmitted into the ship. Secondly, the noise and vibration inside the ship is measured during the sea trial, followed by the ambient noise separation. Finally underwater propeller noise is predicted by using the sound transmission coefficients and measured onboard noise and vibration signals. Variables expressing noise and vibration in the sea and onboard are expressed, and the problems of this prediction method are defined in chapter 2. The proposed methodology and the feasibility test by cavitation tunnel are introduced in chapter 3, followed by the application on actual ships in chapter 4.

2. PROBLEM DEFINITION

There exist several kinds of noises that affect underwater radiated noise level. Among them, the radiated noise from the propeller is known as the most significant noise source. Therefore, we have focused on the prediction of propeller radiated noise in this research.

In order to exactly predict how much propeller noise is radiated to the water by measuring onboard noise, it is essential to understand the process how the propeller radiated noise is transmitted into the ship, and what kind of problems can occur in the prediction. Mathematical expressions help us to define the problems that we want to solve for the accurate prediction. It leads us to define variables expressing the noise inside and outside the ship. Mathematical expressions can be simplified as Eq. (1).
\[ P = T \times P_p + P_r + P_n, \]  

(1)

where \( P \) is the measured onboard noise and \( T \) is the sound transmission coefficient which means how much underwater noise is transmitted into the ship, \( P_p \) is the propeller radiated noise in the water, \( P_r \) is the propeller radiated noise reflected from walls in the ship, \( P_n \) is the measured noises other than propeller noise. The notation for time, frequency and space are omitted for simplicity. Equation (1) means that the propeller radiated noise in seawater is transmitted through the hull structure and reflected by interior walls. The propeller noise is measured together with other noise using the microphone in the ship. Fig. 1 shows conceptual diagram of variables in Eq. (1).

![Fig. 1 Conceptual diagram for onboard propeller noise measurement](image)

Eq. (1) and Fig. 1 indicate that the following problems should be resolved to predict the propeller radiated noise (\( P_p \)) by using measured onboard signals (\( P \)):

i) How to obtain the sound transmission coefficient \( T \)?  
ii) How to remove the reflected noise \( P_r \)?  
iii) How to separate the unwanted noise \( P_n \)?

### 3. PREDICTION OF UNDERWATER RADIATED NOISE IN CAVITATION TUNNEL

This chapter shows the problems issued in Chap. 2 can be solved. In order to check the feasibility of the method, the prediction of propeller radiated noise was carried out using the cavitation tunnel of Hyundai Maritime Research Institute. The underwater noise in the tunnel was predicted by using the measured signals outside the cavitation tunnel, which includes the sound transmission coefficient measurement of cavitation tunnel, separation of ambient noise, and elimination of reflected wave from tunnel wall. The prediction results were compared with the direct measurement results using a hydrophone.
3.1 Sound transmission coefficient of cavitation tunnel

In general, the sound transmission coefficient is dependent on the frequency but is independent on time and type of source signal. This enables us to use noise sources other than propeller to estimate the sound transmission coefficient of cavitation tunnel. In this study, we measured the sound transmission coefficient of cavitation tunnel by using underwater loud speaker which generates broadband noise between 100Hz~10kHz. Fig. 2 shows the experimental setup. Both the hydrophone inside the cavitation tunnel and microphones outside the tunnel measure the noise during the operation of underwater loud speaker. \( H_1 \) estimation [3] is used to reduce the ambient noise in air during the measurement, which is written as

\[
H_1(f) = \frac{S_{12}(f)}{S_{11}(f)},
\]

where \( H_1(f) \) means the sound transmission coefficient of cavitation tunnel. \( S_{11}(f) \) and \( S_{12}(f) \) represents the auto spectrum of signal measured by the hydrophone and cross spectrum between signals by hydrophone and microphones at a frequency \( f \), respectively.

Fig. 2 Experimental setup for sound transmission coefficient measurement of cavitation tunnel

3.2 Separation of ambient noise

The cavitation tunnel used in the experiment is exposed to various noises such as pumps, cranes and other machinery noises, which causes error in the underwater noise prediction. The accelerometer was installed on the outer surface of cavitation tunnel to separate the noises. The basic idea is that the propeller radiated noise is transmitted through cavitation tunnel’s walls whereas other noises are directly radiated to the microphone. Based on this phenomenon, we measured the coherence between microphone and acceleration signal to quantify how much noise measured by microphones is directly related with the propeller radiated noise through the duct wall. Thus the ambient noise can be removed by using Eq. (3)

\[
P_{p,\text{air}}(f) = \gamma^2(f)P_{\text{air}}(f),
\]

where \( P_{p,\text{air}}(f) \) represents the propeller radiated noise transmitted to the air, \( \gamma^2(f) \) is the coherence between microphone and accelerometer on the wall, \( P_{\text{air}}(f) \) is the noise measured by microphones in the air.
3.3 Elimination of reflected wave

The propeller radiated noise in the cavitation tunnel includes reflected sound due to the tunnel wall as well as direct sound from the propeller. In order to estimate the propeller radiated noise, the reflected sound should be eliminated during the prediction process. The simplest way is to compensate the reflection effect by considering the difference of signals between with and without wall. However, it is difficult to build the condition without wall in the water. In this study, we developed the following indirect method:

i) Measurement of underwater speaker noise in the cavitation tunnel filled with water
ii) Measurement of underwater speaker noise in the air with no reflection
iii) Calculation of the characteristic impedance of medium (water and air)
iv) Combination of i)-iii) to derive the relation between sound transmission characteristics with and without reflective wall

This enables to predict the propeller noise level in the free field by measuring the noise in the cavitation tunnel.

3.4 Prediction results

In order to verify the feasibility of proposed methods, cavitation tunnel experiments were carried out. The propeller radiated noise was measured using the hydrophone installed in the cavitation tunnel. At the same time, the transmitted noise through the cavitation tunnel was measured using five microphones to predict the propeller noise. During experiment, three kinds of propeller cavitation were generated and the cavitation noise dominated the measured noise. Fig.3 shows the comparison between prediction and measurement in the three types of cavitation, respectively. The results shows that the proposed method can estimate the propeller radiated noise level with error of about 3dB.

![Fig. 3 Estimation result of cavitation noise and their comparison with measured values](image-url)
4. PREDICTION OF UNDERWATER RADIATED NOISE IN A SHIP

This chapter shows the application results of the proposed method to the prediction of propeller noise level during sea trial for two kinds of ships: one is a crude oil carrier and another is a container carrier.

![Propeller and Rudder](image)

**Fig. 4 Structure-borne noise sensors attached at the propeller and rudder**

4.1 Measurement of sound transmission coefficient of hull structures

Experiments in dry dock were carried out to measure sound transmission coefficient of each ship’s stern. Noises at each ship’s outer spaces, surfaces, and onboard spaces were measured simultaneously while propeller and hull were excited outside by an impact hammer and impulsive noise generator using gunpowder. The hull’s sound transmission coefficients were measured by using the ratio of inside and outside signals.

4.2 Propeller noise measurement during sea trials

In order to predict propeller noise, the structure-borne noises in the ship’s stern side were measured during the sea trial. Fig. 5 shows measured signal at the stern of crude oil carrier, which has impulsive components with a period corresponding to the inverse of blade passing frequency. This shows that the propeller noise induced by cavitation was measured as impulsive sound.

![Signal Measurement](image)

**Fig. 5 Structure-borne noise measured at ship’s stern side of crude oil carrier**

Fig. 6 shows the measured signal at the stern of container carrier. Contrary to the signal measured at crude oil carrier, it is hard to find the periodicity (that is, the existence of propeller cavitation noise) in time domain. The DEMON spectrum [4] was used for the
detail analysis on the propeller noise. DEMON spectrum is a kind of frequency analysis method known as its prominence for the periodic impulse detection. In this paper, DEMON spectrum was utilized to judge whether the measured signal includes cavitation noise. Fig. 7 shows DEMON spectrum of the signal shown in Fig. 6, which as a peak at a frequency corresponding to the blade passing frequency. This implies that the cavitation noise was measured although it is invisible due to measurement noise.

![Fig. 6](image-url)  
Fig. 6 Structure-borne noise measured at the stern side of container carrier

![Fig. 7](image-url)  
Fig. 7 DEMON spectrum of the measured signal at the stern side of container carrier

The measured signal, however, included the noise due to onboard machinery such as engine as well as cavitation noise. In order to separate them, we split the measured signal into a number of time step, and classified them into two groups: impulsive components and non-impulsive regions. By comparing the power spectrum of two groups, we extracted the frequency region where cavitation components mainly contribute.

4.3 Prediction of propeller radiated noise level

By using the extracted spectrum (i.e. the spectrum of cavitation noise) and sound transmission coefficient, the propeller cavitation noises in water were predicted. The predicted values were compared with the averaged underwater noise level of the similar kind of ship [5] as summarized in Fig. 8. Of course, this comparison needs additional verification since the frequency region, type of propeller and cruise condition may not be identical.

5. CONCLUSION

This research proposed a indirect method to predict the propeller radiated noise by onboard measurement. The feasibility test for the proposed method was carried out in the cavitation tunnel experiments, and the method was also applied to prediction of propeller radiated noise from actual ship. In the cavitation tunnel experiments, the underwater loud speaker was utilized to measure sound transmission coefficient of the tunnel wall, and the acceleration signal was measured to reduce the ambient noise. As a result, the proposed method showed to be well agreed with the test result within 3dB error. In the actual ship
The sound transmission coefficient was measured by impact test in the dry dock, and the effect of water was compensated by theoretical approach. The cavitation noise from the ship’s propeller was predicted by using the structure-borne noise at ship’s stern during sea trial and sound transmission coefficient. It is expected that the proposed method may easily enable to predict propeller radiated noise within reasonable error bound.

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SOURCE LEVEL ESTIMATES OF SMALL CARGO SHIPS FROM MEASUREMENTS IN A FJORD

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Abstract: This paper presents estimates of source levels of small cargo ships from measurements conducted in a shallow-water environment of the Oslofjord (Norway). Data were recorded on a hydrophone on the Networked Intelligent Underwater Sensors (NILUS) unit placed on the seabed. Noise due to commercial ships in a nearby shipping lane was measured and corrected to monopole source levels with use of the RAM propagation model. An environmental model was constructed based on input from a prior survey of the measurement area. A vertically distributed source model was used in modeling. Information on ship range and draft was taken from ship AIS data. Estimates of source level spectra (30 Hz – 1.4 kHz) and broadband radiated noise levels are presented and compared with values from the literature.

Keywords: Ship noise, source levels
1. INTRODUCTION

Information on radiated noise levels and source levels of commercial ships and their dependence on ship type and characteristics is of interest in modeling and prediction of global and regional ocean noise [1]. Measurements on commercial ships are typically conducted in shipping lanes [2-3] outside dedicated measurement ranges. This may require data analysis steps not contained in present ship noise measurement standards; this applies in particular for measurements conducted in shallow water [4-5]. In addition, there are relatively fewer reported measurements on small commercial ships typically operating in coastal waters. The objectives of the study reported in this paper were to measure ship noise by use of a rapidly deployable sensor system, and to estimate radiated noise levels and source spectra of commercial ships in a fjord environment.

2. THEORY AND PROCEDURES

The measurement and data processing procedures were set up to follow the specifications for Grade C–Survey Method of the American National Standard (ANSI) [6] where possible. However, experiment considerations forced departures from this standard notably with respect to hydrophone placement (here: at seabed), measurement aspect (here: ±30º of stern) and elevation (here: 8±3º) angles, and multiple runs. Following the ANSI standard, the measured sound pressure level SPL was corrected to radiated noise level RNL using the formula:

\[
RNL = SPL + 10 \log \left( \frac{r^2}{r_{ref}^2} \right) \text{ [dB re } \mu \text{Pa}^2]\]

where \( r \) is the distance from the measurement hydrophone to the ship and \( r_{ref}=1 \text{ m} \) the reference unit. To estimate \( r \) we used the horizontal distance from the hydrophone to the ship position as obtained from ship Automatic Identification System (AIS) data and the vertical distance between the hydrophone and a point source model depth (defined below). SPL was measured from 10 Hz to 1.6 kHz. Hydrophone data, sampled at 9 kHz, were processed in Hanning windowed time segments (length 1 s, 50% overlap), then fast-Fourier transformed (frequency resolution 1 Hz). Signal processing applied Welch power spectral density estimation over a time window equivalent to one ship length (or 100 m) selected within 1-4 minutes of the ship closest point of approach (CPA). Corrections for hydrophone sensitivity and for ambient noise (using a data window of length 60 s within 1 hr of CPA) were applied.

The correction formula in Eq. 1 implicitly assumes a close-distance measurement not influenced by propagation effects such as seabed reflection and scattering, which can be difficult to achieve outside a measurement range. We thus processed data also for monopole source level \( SL^{mp} \) by use of the correction formula [7]:

\[
SL^{mp} = SPL + PL(r) \text{ [dB re } \mu \text{Pa}^2 \text{ m}^2]\]

where \( PL(r) \) is the propagation loss between the source and receiver. This required the use of a suitable acoustic propagation model, described in some detail below.
To compute PL, we used the RAM range-dependent acoustic propagation model [8]. Environmental data (section 3) was sampled at 1-m intervals for sound speed profiles in water, and at 5-m intervals for bathymetry profiles from a gridded model. Based on survey information, a seabed model representative of mud, with sound speed 1502 m/s, attenuation 0.40 dB/λ (acoustic wavelength λ), and density 1.50 g/cm³ was used. The RAM model was run at 100 frequencies logarithmically spaced from 10 Hz to 1.6 kHz. At each frequency, model output intensity was linearly interpolated in range to a grid of 1 m, and then interpolated in frequency to a resolution of 1 Hz. For PL(\(r\)) in Eq. 2 we used a linear average over ±5 m around the ship range at the centre of the data processing window, and a receiver depth fixed at 0.5 m above the seabed.

Two source models [3-4] were tested: (i) a point source at depth \(z_1 = D - 0.85\frac{P}{2}\) with \(D\) the ship draft (obtained from AIS data) and \(P\) the assumed propeller diameter, and (ii) a vertically distributed source modelled as a Gaussian weighted distribution of point sources of width \(\sigma = \frac{P}{4}\) centred on \(z_1\). (Depth sampling of 0.15 m over the 90% highest values of the distribution, truncated at the sea surface, was here used.) Figure 1 shows modelled PL versus receiver range and frequency for \(D=3.5\) m for the two source models, in an infinitely deep waveguide. For the point source model (Fig. 1a) there is a distinct Lloyd Mirror interference pattern at short ranges. For the distributed source model (Fig. 1b), this interference pattern is smoothed; at ranges beyond approximately 500 m the two source models yield comparable PL predictions. Further details of this work can be found in [9]. In the following, the vertically distributed source model was applied.

![Fig.1: Propagation Loss (dB re m²) versus frequency and range for (a) point source and (b) distributed source model for ship draft D=3.5 m in an infinitely deep waveguide.](image)

3. MEASUREMENTS

The measurement area (Figure 2) was in a shallow basin of approximately 12 km width and 6 km breadth in the Oslofjord, Norway. Two shipping lanes cross the area: one north-south toward Oslo, with a western branch towards the port of Drammen. The sensor deployment area was set up to measure noise from ships in both lanes; only data from the shipping lane towards Drammen will be presented here. Two NILUS platforms were deployed on the seabed within the indicated area, at water depths of approximately 110 and 195 m. This lightweight platform, developed at FFI, is designed for stable deployment at the seabed and easy operation from a small workboat [10]. The platforms used in this experiment were equipped with sensors that included a calibrated DIFAR hydrophone (Ultra Electronics) of bandwidth 3 kHz.
Fig. 2: Chart of the Oslofjord experiment area with shipping lanes (solid blue lines), and sensor deployment area (broken blue circle).

Acoustic data were collected five days within the period June 8-24, 2010. AIS data were continuously logged at a nearby shore based receiver. Supporting environmental data consisted of weather information and measurements of salinity-temperature-depth profiles in water, taken daily near the sensor sites. These showed typical summer profiles fairly stable over the duration of the experiment except for variations in the upper 3-5 m attributed to surface heating effects. From survey multibeam sonar data, a gridded bathymetry model had been constructed for the area. This showed a flat area of approximately 200 m water depth to the northwest of the sensor deployment area, and a range-dependent sloping seabed with ridges extending to water depths of 50 m to the east and south of the deployment area. Six seabed grab samples were analyzed for grain size; the range of values of $M_z = 7.9 – 8.7$ indicated mud type sediment.

4. RESULTS

Recorded time series were inspected for ship passages also identified in AIS data. Passages with no other visible passages within 60 minutes were selected for processing. Considerations on bathymetric complexity further led us to consider measurements only on ships in transit to Drammen to the northwest of the sensor deployment area.

Table 1 lists selected ship passages together with ship information obtained from AIS data and additional measurement parameters. The measurements included five ships, measured at ranges of 1100-1400 m and aspect angles within 30° of stern. There were four cargo ships of size 1600-3100 Gross register tonnage (GRT) moving at speeds 10.1-12.2 kts, and one ship (unknown size) identified as a tanker. Table 1 also lists broadband RNL computed using Eq. 1 for 20-1000 Hz. The cargo ships had broadband RNL of 171-177 dB re \( \mu Pa^2 \ m^2 \). The tanker had a broadband RNL of 181 dB re \( \mu Pa^2 \ m^2 \).
Table 1: Ship type, size, length, speed, measurement range and aspect angle (re bow), and estimated broadband radiated noise level RNL in dB re \(\mu Pa^2\ m^2\) (20-1000 Hz).

For comparison, measurements on four large (open hatch) cargo ships of size 20,000-37,000 GRT, length 190-213 m, moving at speeds 13-14.2 kts yielded broadband RNL of 179-184 dB re \(\mu Pa^2\ m^2\) over the same frequency band [2]. The difference in RNL between cargo ships from these data sets is 2-13 dB.

Figure 3 shows estimated source level spectra \(SL_{mp}\) for the four cargo ships of Table 1, computed using Eq. 2 for frequencies 30-1400 Hz. The figure also shows the median spectrum (green curve), and the Wales-Heitmeyer ensemble average spectrum [3] (dashed blue curve). The median spectrum in general follows the Wales-Heitmeyer spectrum, but with lower levels below approximately 100 Hz and higher levels between approximately 300-700 Hz. The rms deviation between the median spectrum and the Wales-Heitmeyer spectrum is 5.1 dB (4.1 dB at 30-340 Hz; 5.4 dB at 340-1200 Hz).

Fig.3: Estimated source level spectra \((SL_{mp})\) of four cargo ships (1600-3100 GRT) measured at near-to-aft aspect in the Oslofjord (black curves), median spectrum (green), and the Wales-Heitmeyer spectrum [3] (dashed blue).
5. SUMMARY AND DISCUSSION

This paper presented estimates of radiated noise levels and source spectra of small commercial ships from measurements conducted in the Oslofjord, Norway. The measurements used a hydrophone on the NILUS lightweight deployable measurement platform placed on the seabed. Broadband radiated noise levels were within 171-177 dB re μPa² m² (20-1000 Hz) for four small cargo ships (1600-3100 GRT) moving at speeds 10-12 kts measured at near-to-aft aspect angles. These levels are 2-13 dB lower than those obtained from measurements on large cargo ships moving at speeds 13-14 kts [2]. Differences in measurement geometry may partly explain this level difference. The estimated source spectra in general followed the Wales-Heitmeyer spectrum [3] but with deviations from, and largest variations among the spectra, below 100 Hz and at 300-700 Hz. These deviations may be attributed to differences in measurement aspect angles, as well as to environmental uncertainty and propagation effects not fully accounted for in the numerical modeling applied for this fjord environment. Further measurements as outlined in this paper are suggested to estimate source levels of a larger number of and other types of small commercial ships.

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Session 18
Sensitivity of underwater acoustic observables

Organizer: Emmanuel Skarsoulis
TRAVEL-TIME SENSITIVITY KERNELS IN A SHALLOW WATER ENVIRONMENT

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Abstract: Travel-time sensitivity kernels of peak and phase arrivals in a Pekeris shallow-water waveguide are considered. Peak arrivals are defined as the maxima of the acoustic pressure envelope at the receiver, whereas phase arrivals are defined as the maxima of the received acoustic pressure itself. While the two-dimensional kernels of peak and phase arrival times are comparable, the three-dimensional kernels exhibit differences in shape and magnitude, which point to differences in the way that the two observables sample changes in the water mass. Phase arrivals have the potential to resolve the sensitivity kernel components of overlapping peak arrivals, whose separation would otherwise require a very large bandwidth.

Keywords: Sensitivity kernels, travel time, phase, shallow water, Pekeris waveguide
1. INTRODUCTION

Sensitivity kernels of underwater acoustic observables reveal the way the water mass is sampled by the different observables. Travel-time sensitivity kernels in particular are the Born-Frechet kernels relating the spatial distribution of sound-speed variations with the induced travel-time variations and rely on the Born approximation for perturbations of the frequency-domain Green’s function [1]. Vertical travel-time sensitivity kernels can be obtained as horizontal marginals of two-dimensional (2D) and three-dimensional (3D) kernels or by direct perturbation of the range-independent Green’s function with respect to the sound speed [1,2].

In this work 2D and 3D travel-time sensitivity kernels of peak and phase arrivals in a Pekeris shallow-water waveguide are considered and compared with each other. Peak arrivals are defined as the maxima of the acoustic pressure envelope at the receiver and they can be modelled in a geometric or wave-theoretic context [1]. The time resolution of peak arrivals is limited by the source bandwidth – the arrival width is inversely proportional to the bandwidth. In this connection, a source of sufficient bandwidth is needed in order to resolve peak arrivals. Phase arrivals on the other hand are the peaks of the acoustic pressure itself and their width is inversely proportional to the source frequency, which is in general much larger than the source bandwidth. Thus, higher temporal resolution can in principle be obtained with phase arrivals than with peak arrivals.

In the following perturbation relations and expressions for travel-time sensitivity kernels are briefly derived and the sensitivity behavior of peak and phase arrivals in a Pekeris shallow-water waveguide is addressed by comparing the corresponding 2D, 3D sensitivity kernels.

2. PERTURBATION RELATIONS AND SENSITIVITY KERNELS

The complex pressure at the receiver in the time domain can be written in the form [3]

\[ p(t) = a(t)e^{j\varphi(t)}e^{j\omega_0 t}, \]  

(1)

where \( t \) denotes time, \( a(t) \) is the amplitude as a function of time (arrival pattern), \( \varphi(t) \) is the phase and \( \omega_0 \) is the central circular frequency of the source. The demodulated pressure

\[ \tilde{p}(t) = a(t)e^{j\omega_0 t} = u(t) + iv(t) \]  

(2)

results after removal of the central frequency and can alternatively be expressed in terms of its real and imaginary parts, \( u \) and \( v \), respectively. The above quantities \( a(t) \), \( \varphi(t) \), \( u(t) \), and \( v(t) \), and the resulting arrival times, depend on the source/receiver location as well as on the sound-speed distribution \( c(x) \), where \( x \) is the spatial variable. Thus, perturbations \( \delta c(x) \) give rise to perturbations in arrival amplitude, phase and corresponding peak and phase arrival times.

Peak arrival times \( \tau_p \) are defined as the times of the local maxima (peak arrivals) of the arrival pattern. The resulting perturbation relation for peak arrival times [3] reads
\[ \delta \tau_p = -\frac{\delta \hat{u}(\tau_p; c; \delta c)}{\hat{u}(\tau_p; c)} = -\frac{u \delta \hat{u} + u \delta \hat{u} + v \delta \hat{v} + v \delta \hat{v}}{u^2 + u \hat{u} + v^2 + v \hat{v}}, \]  
\hspace{1cm} (3)

where the quantities \( a, u, v \) and their time derivatives (dotted), as well as their 1st-order perturbations due to \( \delta c \), are evaluated at the peak arrival time in the background state.

Phase arrival times can be defined on the received pressure as times of constant phase. The phase as a function of time can be written as

\[ \psi(t) = \phi(t) + \omega_d t. \]  
\hspace{1cm} (4)

Due to the term \( \omega_d t \), the phase \( \psi(t) \) is rapidly increasing with time. The resulting perturbation relation for phase arrival times reads [3]

\[ \delta \tau_p = -\frac{\delta \phi}{\phi + \omega_b} = -\frac{u \delta \hat{v} - v \delta \hat{u}}{u \hat{v} - v \hat{u} + \omega_b \left( u^2 + v^2 \right)}. \]  
\hspace{1cm} (5)

The complex pressure \( p \) at the receiver in the time domain, and its perturbation \( \delta p \) due to perturbations of the sound speed distribution, can be expressed through the inverse Fourier transform in terms of the signal \( P_s(\omega) \) emitted by the source in the frequency domain and the frequency-domain Green’s function \( G_{sr}(\omega; c; x_s|x_s) \) and its perturbation \( \delta G_{sr}(\omega; c; \delta c; x_s|x_s) \), respectively,

\[ p(t) = \frac{1}{2\pi} \int_{-\infty}^{+\infty} P_s(\omega)G(\omega; c; x_s|x_s)e^{j\omega t}d\omega, \quad \delta p(t) = \frac{1}{2\pi} \int_{-\infty}^{+\infty} P_s(\omega)\delta G(\omega; c; \delta c; x_s|x_s)e^{j\omega t}d\omega, \]  
\hspace{1cm} (6)

where \( x_s \) and \( x_r \) is the source and receiver position vector. The perturbation of the Green’s function can be expressed through the first Born approximation [2], which in 3-dimensional (3D) space reads

\[ \delta G_{3D}(\omega; c; \delta c; x_s|x_s) = -2\omega^2 \iiint_V G_{3D}(\omega; c; x_s|x_s)G_{3D}(\omega; c; x_s|x_s) \frac{\delta c(x)}{c^3(x)}dV(x), \]  
\hspace{1cm} (7)

where \( V \) is the volume spanned by the 3D position vector \( x \) and \( dV \) the corresponding volume differential. Assuming a range-independent background environment the 3D Green’s function in the far field can be written in terms of normal modes [2]

\[ G_{3D}(r, z | z_s) = \frac{e^{-j\sigma/4}}{\sqrt{8\pi}} \sum_{m=1}^{M} \frac{\varphi_m(z_s)\varphi_m(z)}{\sqrt{k_m r}} e^{-ik_m r}, \]  
\hspace{1cm} (8)

where \( k_m \) and \( \varphi_m(z) \) are the real eigenvalues and the corresponding eigenfunctions of the vertical Sturm-Liouville problem. By substituting eq. (8) into eq. (7) and using the perturbation relations, eqs. (3) and (5), expressions of the form

\[ \delta \tau = \iiint_V K_{\tau}^{(3D)}(x) \delta c(x)dV(x) \]  
\hspace{1cm} (9)

can be obtained, where \( K_{\tau}^{(3D)}(x) \) is the corresponding 3D travel-time sensitivity kernel, describing the effect that a sound-speed perturbation at location \( x \) will have on the travel time \( \tau \) of interest (either peak or phase arrival time). A similar expression can be obtained for the 2-dimensional (2D) sensitivity kernel based on the 2D Green’s function [2]
\[
\delta \tau = \int_A K^{(2D)}_r(x) \delta c(x) dA(x),
\]

where \(A\) is the area spanned by the 2D position vector \(x\) and \(dA\) the corresponding area differential.

3. NUMERICAL RESULTS

Some numerical results are presented in the following for travel-time sensitivity kernels of peak and phase arrival times in a Pekeris shallow-water waveguide shown in Fig. 1. The water depth is 200 m and the sound speed in the water is taken 1500 m/s. The bottom is taken to be a half space of sound speed 1900 m/sec. Source and receiver are at a depth of 50 m – marked by the dots in Fig. 1 – and horizontal distance of 3 km. The acoustic signal is a Gaussian pulse with central frequency 300 Hz and bandwidth 70 Hz (3-dB bandwidth).

Fig. 2 shows at the top the 2D arrival pattern on the left and the acoustic pressure at the receiver in the time domain (real part) on the right. The lower panels show the 2D travel-time sensitivity kernels corresponding to the 3 marked peaks (peak arrival times on the left and phase arrival times on the right).

Fig. 2. 2D propagation results in the Pekeris waveguide at 3 km range. Left: Arrival pattern and sensitivity kernels for peak arrival times of marked peaks. Right: Acoustic pressure and sensitivity kernels for phase arrival times of marked peaks.
arrival pattern starts with near-horizontal arrivals corresponding to low-order modes and continues with gradually steeper arrivals, with increasing separation, until the critical angle is reached. The real part of the received pressure on the right shows a modulated version of the arrival pattern with modulation frequency corresponding to the central source frequency, 300 Hz. The 3 marked phase arrivals on the right are selected to be closest – in time – to the selected 3 peak arrivals on the left.

The sensitivity kernel of the first peak arrival (peak 1) represents the superposition of two propagation paths, one surface-reflected and one bottom-reflected, interfering in the time domain. The underlying arrivals and corresponding propagation paths can be resolved by increasing the bandwidth (Fig. 5 below). The sensitivity kernel of the first phase arrival corresponds to the surface reflected path. The arrival corresponding to the direct (horizontal) path cannot be resolved / separated from the surface-reflected arrival, because the difference between the corresponding path lengths is very small (1.7 m) resulting in a difference in arrival times of 1.1 msec. The kernels of the later arrivals correspond to steeper acoustic paths with larger number of surface and bottom reflections. Peak and phase travel-time sensitivity kernels for those arrivals are comparable in shape and magnitude.

Fig. 3 shows the 3D propagation results for the arrival pattern and peak arrival time sensitivity kernels – intersection with the vertical plane passing through the source and receiver – on the left and the received acoustic pressure and phase arrival time sensitivity kernels on the right. While the arrival pattern and acoustic pressure are very similar to the 2D results of Fig. 2 the 3D sensitivity kernels of the peak and phase arrivals exhibit a large
Fig. 4. 3D propagation results in the Pekeris waveguide at 3 km range. Left: Arrival pattern and sensitivity kernel cross-sections at mid range for peak arrival times of marked peaks. Right: Acoustic pressure and sensitivity kernel cross-sections for phase arrival times of marked peaks.

Fig. 5. 2D propagation results in the Pekeris waveguide using a bandwidth of 210 Hz. Left: Arrival pattern and peak arrival time sensitivity kernels for 3 marked peaks. Right: Received acoustic pressure and phase arrival time sensitivity kernels for 3 marked peaks.
difference in magnitude, with the peak arrival time sensitivity being much weaker than the sensitivity of phase arrival times. This difference points to a dissimilar off-plane behavior of the 3D kernels for the phase and peak arrivals.

Fig. 4 shows the vertical cross-section (cross-range direction) of the 3D peak and phase arrival time sensitivity kernels at 1.5 km range, i.e. in the middle between source and receiver. From the right-hand panels of this figure it is seen that the sensitivity of phase arrival times exhibits near-axial symmetry about its core along with a strong alternating behavior, which causes cancelation of the central high sensitivity values when horizontal cross-range marginals are taken. On the other hand the negative 3D sensitivity of the peak arrival times (left) extends in the horizontal cross-range direction and this causes an amplification of the central values. This indicates that the two observables sample changes in the water mass differently, phase arrival times within the Fresnel zone about the eigenrays and peak arrival times in the horizontal cross-range direction, sideways from the eigenray.

Fig. 5 shows higher-resolution propagation results and sensitivity kernels resulting from a triple-fold increase in the signal bandwidth (210 Hz). This leads to a corresponding decrease in the pulse duration such that overlapping peak arrivals can be resolved as can be seen in the case of the congested early arrivals (near-horizontal propagation paths). In that case the first peak in Fig. 2 is the superposition of the first two peak arrivals in Fig. 5, and its sensitivity kernel splits into two sensitivity kernels one corresponding to the surface-reflected and one to the bottom-reflected path, peaks 1 and 2 in Fig. 5, respectively. The time difference between these two arrivals is 10 msec and can be resolved with the 210-Hz bandwidth (5-msec pulse duration), but not with a 70-Hz bandwidth (14-msec pulse duration).

The third peak arrival in Fig. 5 is a double arrival corresponding to two symmetric propagation paths, with one surface and one bottom reflection each. Coming to the sensitivities of the phase arrival times, the situation is similar as for the peak arrivals. The interesting point in the case of phase arrivals is that in order to resolve the two different propagation paths, the surface- and the bottom-reflected one, it is not necessary to increase the bandwidth. Fig. 6 shows at the top a detailed view of the early part of the 70-Hz arrival pattern shown in Fig. 2 and three phase arrivals selected, with times close to those of the selected three arrivals in Fig. 5. The sensitivity kernels shown in the lower panels of Fig. 6 are practically the same as the ones obtained with a triple-fold bandwidth in Fig. 5. This is a significant advantage of phase arrivals and the corresponding travel time sensitivity kernels.

Fig. 6. Received acoustic pressure (detail) and phase arrival time sensitivity kernels for 3 early phase arrivals (marked) in the Pekeris waveguide using a bandwidth of 70 Hz.
4. CONCLUSIONS

Two- and three-dimensional travel-time sensitivity kernels of peak and phase arrivals in a Pekeris shallow-water waveguide were presented. While the 2D kernels are comparable, the 3D kernels exhibit differences in shape and magnitude, which points to differences in the way that the two observables sample changes in the water mass. Cross-sections of the 3D kernels reveal that the sensitivity of phase arrival times exhibits near-axial symmetry about the corresponding eigenrays, whereas that of the phase arrival times extends in the horizontal cross-range dimension, i.e. sideways from the eigenrays. Phase arrival times offer higher temporal resolution than peak arrivals, since their width is controlled by the central source frequency rather than the source bandwidth. Further, it was shown that phase arrivals can resolve the sensitivity kernel components of overlapping peak arrivals whose separation would otherwise require a very large bandwidth.

5. ACKNOWLEDGEMENTS

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A HYBRID APPROACH FOR OCEAN ACOUSTIC TOMOGRAPHY
BASED ON STATISTICAL CHARACTERIZATION OF THE
ACOUSTIC SIGNAL AND THE IDENTIFICATION OF MODAL
ARRIVALS

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Abstract: A hybrid approach is presented for problems of ocean acoustic tomography, based on the statistical characterization (SC) of the acoustic signal. The statistical characterization is used for the estimation of a reference solution to the inverse problem of estimating the sound speed profile in the water column using a Genetic Algorithm. By applying first order perturbation approach, variations of the sound speed profile are associated with modal travel time variations. This relationship provides the framework for the development of an iterative scheme which converges when the reference environment is close to the actual one and provides a fine tuning of the results obtained by the original method. The performance of the method is demonstrated by means of a simulated experiment in range-independent environment.

Keywords: Signal Processing, Sensitivity Kernels, Inverse Problems
1. INTRODUCTION

This paper deals with a hybrid inversion scheme for the estimation of the sound speed profile in shallow water environment, based on the exploitation of an acoustic signal received at a single hydrophone. The signal is initially characterized using a statistical approach which, with the help of a Genetic Algorithm (GA) leads to an initial estimation of the environmental parameters under consideration [1]. The inversion results are further improved by means of a linear inversion scheme, based on mode identification and the use of a linear sensitivity kernel introduced by Rajan et al [2] and thereafter applied for tomographic inversions either as a stand alone inversion [3] or in a second phase of a hybrid inversion scheme [4]. Actually the scheme suggested here is similar to the one presented in [4], with the only difference being the first phase, which, instead of applying a matched modal arrivals scheme as in [4], we apply a statistical characterization scheme associated with the GA.

The scheme is tested here for the case of a shallow water range independent environment consisting of the water column and the sea-bed formed with a sediment layer over a semi-infinite half space. The sound speed in the water and sediment is considered to vary linearly with depth, whereas the sound velocity in the substrate is constant. The inversions will be performed here for the estimation of the sound speed profile in water. A Gaussian source is assumed and the proposed approach will be tested using noisy synthetic data (SNR 17 dB). Table 1 presents the geoaoustic parameters of the environment as well as the operational details of the source, along with the original search space of the recoverable parameters.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Actual Value</th>
<th>Lower bound</th>
<th>Upper bound</th>
</tr>
</thead>
<tbody>
<tr>
<td>Receiver range $r(m)$</td>
<td>9000.0</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Source central frequency $f_c(Hz)$</td>
<td>120.0</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Source bandwidth $\Delta f(Hz)$</td>
<td>45.0</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Source depth $z_s(m)$</td>
<td>60.0</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Receiver depth $z_r(m)$</td>
<td>50.0</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Water depth $h(m)$</td>
<td>100.0</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Sound speed at the surface $c(0)(m/s)$</td>
<td>1490.0</td>
<td>1485.0</td>
<td>1500.0</td>
</tr>
<tr>
<td>Sound speed at the w/s interface $c(h)(m/s)$</td>
<td>1500.0</td>
<td>1490.0</td>
<td>1520.0</td>
</tr>
<tr>
<td>Sediment thickness $d(m)$</td>
<td>40.0</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Sediment sound speed $c_s(h)(m/s)$</td>
<td>1550.0</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Sediment sound speed $c_s(h+d)(m/s)$</td>
<td>1600.0</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Sediment density $\rho_s(kg/m^3)$</td>
<td>1600.0</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Substrate sound speed $c_{sb}(m/s)$</td>
<td>1800.0</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Substrate density $\rho_{sb}(kg/m^3)$</td>
<td>1800.0</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Table 1: The environment of the test case
Figure 1 presents the simulated signal obtained using the Normal Mode program MODE 1 to calculate the system transfer function $H(x_s,x_r;\omega)$ where $x_s$ and $x_r$ are the source and receiver positions vectors respectively. The system transfer function is thereafter multiplied by the source excitation function $S(\omega)$, and by means of an Inverse Fourier transform we get the signal in the time domain.

$$p(x_s,x_r;\omega) = H(x_s,x_r;\omega)S(\omega)$$

$$S(x_s,x_r;t) = \mathcal{F}^{-1}[p(x_s,x_r;\omega)]$$

The signal is characterized using the statistics of the wavelet sub-band coefficients as suggested in [1]. In previous works it has been shown that the wavelet coefficients of a typical tomographic signal, here simulated by means of a Gaussian excitation function $S(\omega)$, obey a Symmetric Alpha Stable distribution (SaS) defined by a pair of the parameters $(\alpha, \gamma)$.

Thus, for an L-level wavelet analysis, the signal can be characterized by L detailed and 1 approximation coefficient vectors $\Phi$, each one of which consisting of only two elements. Hence the signal feature is represented by a vector $d$ as following:

$$S(x_s,x_r,t) \leftrightarrow \{\Phi_0,\Phi_1,\ldots,\Phi_L\} \leftrightarrow d = \left[\alpha_0,\gamma_0,\alpha_1,\gamma_1,\ldots,\alpha_L,\gamma_L\right]^T$$

It was also shown that $L = 3$ is an adequate limit of the multilevel analysis of typical underwater tomographic acoustic signals. Thus, the observables of a typical reception of a tomographic signal consists of a vector of 8 elements.

The inverse problem is therefore defined as following:
Given a vector \( \mathbf{d} \) of the statistical coefficients of the wavelet transformed version of the signal, estimate the model parameters \( \mathbf{m} \), which are associated with the signal statistical observables by means of a non-linear relationship:

\[
T(\mathbf{d}, \mathbf{m}) = 0
\]

(3)

The inverse problem is non-linear and is amenable to optimization inversion schemes, based on repeated simulations of the received signal for a class of candidate environments and the use of a cost-function which is appropriate for the nature of the observables. Here, the fact that the actual observables are statistical distributions, an efficient cost-function is the Kullback-Leibler Divergence (KLD) \([5]\) which can be written in a closed form relation:

\[
D_2(S_1, S_2) = \sum_{k=0}^{L} \ln \left( \frac{c_k^{1}}{c_k^{2}} \right) - 1 + \frac{\gamma_2^{k}}{\gamma_1^{k}} \frac{\Gamma \left( \frac{\alpha_2^{k} + 1}{\alpha_1^{k}} \right)}{\Gamma \left( \frac{1}{\alpha_1^{k}} \right)}
\]

(4)

Where \( \Gamma(x) \) is the Gamma function and

\[
c_i^k = \frac{2\Gamma \left( \frac{1}{\alpha_i^k} \right)}{\alpha_i^k \gamma_i^k}, \quad i = 1, 2, \quad k = 0, ..., 3
\]

(5)

The optimization process is performed by using a Genetic Algorithm (GA) with a specific set of algorithm parameters.

For the case under consideration, the vector \( \mathbf{m} \) consists of two elements, corresponding to the sound velocity at the surface and the water-sediment interface: \( \mathbf{m} = [c(0), c(h)]^T \). Remember that we have assumed that the sound speed varies linearly with depth.

The inversion scheme will provide a solution denoted by \( \hat{\mathbf{m}} \) which, in the case of the GA optimization scheme can be chosen to correspond to the best individual of the final population.

3. LOCAL SEARCH SCHEME

The scheme described above is not based on any specific physical observable as it is the case of traditional inversion schemes. This is actually its basic advantage. However, when physical observables can be defined in the recorded signal, alternative inversion schemes can be applied. The idea of a hybrid scheme already suggested in \([4]\) seems to be a good supplement to the non-linear inversions, leading to a fine tuning of the (already good) inversion results. Linear inversion schemes are based on a linear relationship between variations of the recoverable parameters with respect to a reference environment and variations of observables between the actual signal and a simulated signal for the reference environment. Linear inversion schemes are usually applied in an iterative sense the convergence of which is based on the selection the reference environment which has to be as close as possible to the actual one.
In our approach, the observables are the modal arrivals. The modal arrival times \( t_n \) are defined on the basis of the group velocity \( v_{g,n} \) for a specific mode \( n \) with associated eigenvalue \( k_n \):

\[
    t_n = \frac{r}{v_{g,n}}, \quad v_{g,n} = \frac{\partial \omega}{\partial k_n} |_{\omega_0}
\]

(6)

Given a reference environment characterized by a sound speed profile \( c_0(z) \), the tomographic signal for a given source can be simulated giving arrival times denoted as \( t_0^n \). The actual environment corresponds to a sound speed profile \( c(z) \). The perturbation of the sound speed profile is \( \delta c(z) = c(z) - c_0(z) \). Similarly the perturbation of the arrival times of the propagating modes is \( \delta t_n = t_n - t_0^n \). Using expressions (6) to associate the travel time and sound speed differences and expressing the eigenvalue differences with respect to the sound speed differences as in [4], we get the expression:

\[
    \delta t_n = t_n - t_0^n = \frac{\partial \delta k_n r}{\partial \omega} = -\frac{\partial}{\partial \omega} \left( r \int \frac{1}{k_n^2(z)} \rho(z) |u_n^0(z)|^2 \frac{(k_0^2 \delta c(z))^2}{c_0(z)} dz \right)
\]

(7)

This expression provides the kernel of the travel time perturbations with respect to the sound speed perturbation. It should be noted that the expression is based on a first order perturbation approach. Speaking with terms of inverse problems, this expression defines a continuous inverse problem on the \( \delta c(z) \), when measurements of the modal travel time perturbations are available. To facilitate the inversion we transform this problem into a discrete inverse problem using appropriate expansion of the sound speed profile, if a-priori information of its structure is available. In our case, since we have assumed from the first phase that the sound speed varies linearly with depth, we can express the sound speed variation in the water column in terms of appropriate Empirical Orthogonal Functions (EOFs) as following:

\[
    \delta c(z) = A_1 \Phi_1(z) + A_2 \Phi_2(z),
\]

(8)

where, \( \Phi_1(z) = z \), \( \Phi_2(z) = \frac{2h}{3} - z \) and \( h \) is the water depth.

The sound speed profile is therefore determined by means of the vector \( \delta a = [A_1, A_2]^T \).

Introducing the expression for \( \delta c(z) \) from (8) in (7), and performing the integration, we result in a linear system of equations on the unknown vector \( \delta a \):

\[
    \delta t = Q_0 \delta a.
\]

(9)

The kernel \( Q_0 \) is a \( N \times 2 \) matrix, where \( N \) is the number of identified modes calculated for the reference environment.
The system being overdetermined is solved by Singular Valued Decomposition and the coefficients of the EOFs are estimated. Then the sound speed profile is estimated as following:

\[ c(z) = c_0(z) + \delta c(z) = c_0(z) + (A_1 \Phi_1(z) + A_2 \Phi_2(z)). \]  

(10)

We then start an iterative scheme, where the sound speed profile determined in (10) is treated as the reference environment for a new application of the whole scheme. In each step \( m \) of the scheme, a new sound speed profile is estimated:

\[ c_m(z) = c_{m-1}(z) + \delta c_m(z), \quad m = 1, ..., M \]  

(11)

In our notation, \( 0 \) denotes the initial reference environment, and \( M \) is the maximum number of iterations defined on the basis of the convergence criterion:

\[ \| \delta c \| \leq \varepsilon \]  

(12)

where, \( \| \delta c \| = ( (c_m(0) - c_{m-1}(0))^2 + (c_m(h) - c_{m-1}(h))^2 )^{1/2} \) and \( \varepsilon \) is a predefined number.

It should be pointed out that in order that the scheme is applicable, \( N \geq 2 \) modal arrivals should be defined. The identification of the modal arrivals is a non-trivial task. The identification can be obtained automatically as suggested in [6], or manually by applying normal-mode analysis in the signal reproduced for the inversion results of the first phase (GA optimization) and excluding from the identification low energy peaks. It is not the purpose of this work however to focus on the modal identification scheme.

4. INVERSION RESULTS

The hybrid scheme is applied in the environment presented in Table 1 and for the operational characteristics mentioned in the same table. As described above, our inversion procedure consists of two phases. The first phase corresponds to the inversion via Statistical Characterization Scheme (SCS), in which there is no need for any physical observable identification and the second phase corresponds to the linear inversion scheme based on modal arrivals identification.

Thus, we start from the search space for the sound speed in the water column as defined in Table I and the simulated signal presented in Figure 1. By means of the SCS assisted by the KLD Divergence (4) and the GA after 30 Generations with crossover probability 0.8 and mutation 0.02 we come up with the a-posteriori statistical distributions of the final population of the GA presented in Figure 2, where the EOFs have been transformed to sound speed values.
The differences between the best individual (denoted by cross) and the actual values for the sound speed in the surface and bottom (denoted by star) are clearly seen there. Using these values, we simulate the signal in the time domain which is very close to the actual one in terms of travel times of the individual modes as seen in Figure 3. We then proceed with the identification of the modal arrivals. Seven (7) modal arrivals are identified and the differences between actual and estimated travel times (being the input to the linear inversion scheme of the second phase) are defined. The 7 modal arrivals correspond to modes (1st, 7th, 9th, 15th, 17th, 19th and 22nd).

Starting an iterative process as described above, we come after just 4 iterations to the results appearing in Table 2. The recovered sound speed values at the surface and bottom are very close to the actual ones, with differences just in the second decimal point.

Figure 2: A-posteriori statistical distributions of the final population of GA. Horizontal axis corresponds to sound speed in m/sec

Figure 3: The normalized amplitudes of the actual signal and the signal simulated using the inversion results from the GA
5. CONCLUSIONS

We presented here a two-phase hybrid inversion scheme for the estimation of the sound speed profile in the water column, based on the measurement of the acoustic field due to a known source at a single hydrophone. The second phase of the scheme is based on a first order sensitivity kernel. The physical observables are the modal arrivals which should be identified in the signal. A non-linear inversion scheme based on a statistical characterization of the acoustic signal is used in the first phase. Using synthetic noisy data in a shallow water environment it has been demonstrated that the suggested approach leads to excellent inversion results for a sound speed profile varying linearly with depth. Future works include applications in range-dependent environments and the use of real data.

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NON-PERTURBATIVE EVALUATIONS OF TIME SENSITIVITY KERNELS USING ALTERNATIVE DEFINITIONS OF PROPAGATION DELAY

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Abstract: The Time Sensitivity Kernel (TSK) is a classical measure of the observability of sound speed fluctuations from time delays; it is generally derived using the perturbative approximations of Born, Rytov or Keller-Rytov. An intellectually bothering property of the TSK is the “doughnut banana paradox”: in 3D, the TSK is zero just along the eigenray connecting source and receiver, and features maximum weight in a tube around this ray. Recent claims attribute this paradox to Born or Rytov approximations as only first order terms, to be “corrected” by better evaluation techniques or definitions of time. For investigating this question, we evaluate exactly the TSK, using three different definitions of travel time: 1) location of maximum correlation with transmitted waveform; 2) centre of mass of the squared pulse envelop (pertinent for passive configuration; 3) “instantaneous time” related with the derivative of the phase, and pertinent for a waveform with a flat spectrum over a large part of its support. Fréchet derivatives of time delays upon sound-speed fluctuations according to these alternative definitions are evaluated, with no further approximation. The different three definitions of time delay produce highly similar, but different expressions for the TSK, depending not only on mean frequency, but also on the various parameters of the waveform: pulse length, bandwidth... In single eigen-ray configurations, all these TSK feature the “doughnut banana” structure, with cancellation of TSK along the mean ray. Our analysis based on exact evaluation of the TSK confirms the common opinion based on perturbative expressions of TSK.

Keywords: sensitivity kernel, travel time, time delay, doughnut-banana paradox.
1. VARIATIONAL DERIVATIVE OF AN ACOUSTIC FIELD

Variational (or functional) and Fréchet derivatives are extensions to the concepts of ordinary derivatives or partial derivatives (see e.g. Sobczyk [1], pp.31-37). They measure the relative variation \( \delta \Phi \) of an output functional \( \Phi[f] \) to a perturbation \( \delta f(x) \) of the input function \( f(x) \) around some point \( x \). One may consider a sequence of perturbations \( \delta f_n \), all including the same point \( x \), with smaller and smaller supports (i.e. with diameters \( \Delta_n \) tending toward 0); the ratio of the resulting \( \delta \Phi_n \) to the total integral of the perturbation \( \delta f \) may feature an unique limit, independent on the exact shapes of the sequence of perturbations \( \delta f \). This limit, if it exists, is the variational derivative of \( \Phi[f] \) with respect to \( f \), around point \( x \), and is denoted \( \frac{\delta \Phi[f]}{\delta f(x)} \):

\[
\frac{\delta \Phi[f]}{\delta f(x)} = \lim_{\Delta \to 0} \frac{\{\Phi[f + \delta f] - \Phi[f]\}}{\int \delta f(x') dx'}
\]

The variational derivative may be more practically defined as the kernel of a linearized representation of the output \( \Phi[f + \Delta f] \) in terms of the input perturbation \( \Delta f \):

\[
\Phi[f + \Delta f] = \Phi[f] + \int \delta f(x) \frac{\delta \Phi[f]}{\delta f(x)} + O[\Delta f^2]
\]

Variational derivatives may be invoked for quantifying how an acoustic wave, propagating through an inhomogeneous medium, is sensible on variations \( \delta c(r) \) of the sound speed distribution. We consider the acoustic field radiated by a monopole source located at \( s \). The Green function of the problem is governed by the transient wave equation:

\[
\Delta_x G - \frac{1}{c^2(r)} G = \delta(t-s) \delta(t)
\]

The time Fourier transform of this Green function is solution to the harmonic Helmholtz equation:

\[
\omega^2 \frac{\delta G(r,s,\omega)}{\delta c(r_0)} G(r,s,\omega) = \delta(r-s)
\]

Deriving variationally this Helmholtz equation with respect to sound-speed \( c \) provides an equation for the variational derivative of \( G(r,s,\omega) \) with respect to sound speed \( c \) around location \( r_0 \):

\[
\frac{\Delta_x}{c^2(r)} \frac{\delta G(r,s,\omega)}{\delta c(r_0)} + \frac{\omega^2}{c^2(r)} \frac{\delta G(r,s,\omega)}{\delta c(r_0)} = 2 \frac{\omega^2}{c^3(r_0)} \delta(r-r_0) G(r_0,s,\omega)
\]

This last equation is the Helmholtz equation with a Dirac-like source term; the solution may then be written in term of the Green function \( G \):

\[
\frac{\delta G(r,s,\omega)}{\delta c(r_0)} = 2 \frac{\omega^2}{c^3(r)} G(r,r_0,\omega) G(r_0,s,\omega)
\]

The transmitted signal is the waveform \( s_0(t) \), with Fourier transform \( \sigma_0(\omega) \):
The received signal at location \( r \) is the time-convolution of this waveform with the Green function:

\[
 s(t) = \int dt' s_0(t') G(r, s_r, t - t') = \frac{1}{2\pi} \int d\omega e^{-i\omega t} \sigma_0(\omega) G(r, s_r, \omega)
\]

(3)

Derived variationally with respect to sound-speed, these last expressions gives an integral formula for the derivative of the received signal with respect to sound speed:

\[
 \frac{\delta s(t)}{\delta c(r_0)} = \frac{1}{2\pi} \int d\omega e^{-i\omega t} \sigma_0(\omega) \frac{2\omega^2}{c^2(t)} G(r, r_0, \omega) G(r_0, s_r, \omega)
\]

(4)

This expression must be completed with a pertinent form for the Green function corresponding to the unperturbed environment. For example, if the reference unperturbed medium is unbounded, with constant sound-speed \( c_0 \), the harmonic Green function is:

\[
 G(r_1, r_2, \omega) = -\frac{1}{4\pi} \frac{e^{i\omega |r_1 - r_2|}}{|r_1 - r_2|}
\]

(5)

We are then able to derive equations for any function or functional depending on the received signal; such a function is the travel time \( T \), some pertinent definition of which has now to be given. The variational derivative \( \delta T/\delta c(r) \) is the so-called Time Sensitivity Kernel (TSK) and measures the observability of sound speed fluctuations from the corresponding changes of time delay. In the following sections, we will consider three different alternative definitions of travel time.

2. MAXIMUM CROSS-CORRELATION OF RECEIVED SIGNAL WITH TRANSMITTED WAVEFORM

In a tomographic or active sonar configurations ([2]), the transmitted signal \( s_0 \) is known, and the travel time \( T \) may be defined quite naturally as the location in time of the maximum cross-correlation \( S \) of the received signal \( s \) with the transmitted waveform \( s_0 \) (adapted filtering; see e.g. Carter [3]):

\[
 S(t) = \int dt' s(t') s_0^*(t' - t) = \frac{1}{2\pi} \int d\omega e^{-i\omega t} |\sigma_0(\omega)|^2 G(r, s_r, \omega)
\]

where \( \ast \) denote the conjugate of the complex number \( x \). The variational derivation of this last definition immediately gives the derivative of cross-correlation \( S \) with respect to \( c \):

\[
 \frac{\delta S(t)}{\delta c(r_0)} = \frac{1}{2\pi} \int d\omega e^{-i\omega t} |\sigma_0(\omega)|^2 \frac{2\omega^2}{c^2(t)} G(r, r_0, \omega) G(r_0, s_r, \omega)
\]

(6)

The time delay \( T \) maximizes the squared module of the cross-correlation:

\[
 \frac{d\Sigma}{dt} = 0 \quad \text{where} \quad \Sigma(t) = |S(t)|^2 = S(t)S^*(t)
\]

Varational derivative yields:

\[
 \frac{\delta T}{\delta c(r_0)} \frac{d^2\Sigma}{dt^2}\bigg|_T + \frac{\partial}{\partial t} \frac{\delta \Sigma(t)}{\delta c(r_0)} = 0
\]

i.e. finally an expression for the TSK, involving the derivative (6) of cross-correlation:

\[
 \frac{\delta T}{\delta c(r_0)} = -\frac{1}{d^2\Sigma\bigg|_T} \frac{\partial}{\partial t} \frac{\delta \Sigma(t)}{\delta c(r_0)} = 0 \quad \text{with} \quad \frac{\delta \Sigma(t)}{\delta c(r_0)} = 2 \text{Re} \left\{ \frac{\delta S(t^*)}{\delta c(r_0)} S^*(t) \right\}
\]

(7)
As a typical example of wide-band signal, we will model the waveform as a sinus-cardinal with a rectangular Fourier transform, centred around the mean frequency \( \omega_0 \) and with bandwidth \( \Omega \). We will otherwise make an intensive use of the following notations:

\[
\begin{align*}
\tau &= \frac{|r-s|}{c_0} \\
\tau_0 &= \frac{|r-r_0| + |r_0-s|}{c_0}
\end{align*}
\]

One may observe that \( \tau \) and \( \tau_0 \) are identical when \( r_0 \) just lies on the line connecting the source \( s \) and the receiver \( r \). The inequality \( \tau \leq \tau_0 \) always holds.

Over the background of an uniform unbounded environment (using the form (5) of the Green function), the TSK may be fully evaluated after tedious calculations:

\[
\frac{\delta T}{\delta c(r_0)} = -3 \frac{|r-s|}{|r-r_0|} \frac{\Omega}{c_0^3} \left[ A \cos(\omega_0(\tau - \tau_0)) - B \sin(\omega_0(\tau - \tau_0)) \right]
\]

where:

\[
A = \left( \frac{\omega_0}{\Omega} \right)^2 \sin c\left( \frac{1}{2} \Omega(\tau - \tau_0) \right) - \frac{1}{4} \sin c\left( \frac{3}{2} \Omega(\tau - \tau_0) \right)
\]

and:

\[
B = \frac{\omega_0}{\Omega} \sin c\left( \frac{1}{2} \Omega(\tau - \tau_0) \right)
\]

3. TIME DELAY DEFINED AS THE CENTRE OF MASS OF RECEIVED SIGNAL

A rough, but robust definition for the time delay may be the centre of mass of the signal amplitude or squared amplitude ([4]):

\[
T = \frac{\int dt |s(t)|^2}{\int dt |s(t)|^2}
\]

This definition gives the location in time of the “middle” of the signal envelop, provided the transmitted waveform does not feature too sharp and complex amplitude variations. Deriving this definition variationally with respect to \( c \) and using the definition itself of the delay finally give the following expression of the TSK:

\[
\frac{\delta T}{\delta c(\xi_0)} = \frac{1}{\int dt |s(t)|^2} \int dt \left( t - T \right) \frac{\delta |s(t)|^2}{\delta c(\xi_0)} \text{ with } \frac{\delta |s(t)|^2}{\delta c(\xi_0)} = 2 \text{Re} \left\{ \frac{\delta s(t)}{\delta c(\xi_0)} s^*(t) \right\}
\]

Over the background of an uniform unbounded environment (using the form (5) of the Green function), and with expression (4) of \( \delta s/\delta c(\xi_0) \), we calculate the TSK:
As a typical example complementary to the case considered in the previous section, we will adopt a narrow band signal with frequency \( \omega_0 \) and duration L.

\[
\frac{\delta T}{\delta c(r_0)} = \left| \frac{r-s}{r-r_0} \right| \frac{1}{c_s^2} \left[ \int dt (t-T) \text{Re} \left\{ \omega_0^2 (t-\tau_0) s^*(t-\tau) \right\} \right]
\]

where the function f is defined as follows:

\[
f(X) = \begin{cases} 
0 & \text{if } X < -1 \\
\frac{1}{2} X(1+X) & \text{if } -1 \leq X < 0 \\
\frac{1}{2} X(1-X) & \text{if } 0 \leq X < 1 \\
0 & \text{if } 1 \leq X
\end{cases}
\]

4. “INSTANTANEOUS TRAVEL TIME” (RELATIVE DERIVATIVE OF UNWRAPPED PHASE)

If the transmitted waveform was a Dirac pulse \( \delta(t) \), the Fourier transform of the received signal would have the following form: \( \sigma = A e^{i\omega T} \), and the ratio \( 1/\sigma \text{ d}\omega/\text{d}\omega_0 \) would be exactly equal to \( i \) times the time delay T. By analogy with this very particular limit case, the travel time T may be more generally defined as follows ([6], [7]):

\[
T = \text{Im} \left\{ \frac{1}{\sigma(\omega)} \frac{\text{d}\sigma}{\text{d}\omega_{\text{lo}}} \right\}
\]

This definition is pertinent for the specific class of sharp pulses featuring a Fourier transform, which is constant over a significantly large part of its support; the frequency \( \omega_0 \), where the definition applies, must be located over this part of the support.

The Fourier transform \( \sigma \) of the signal and its variational derivative are:

\[
\sigma(\omega) = \sigma_0(\omega) G(r,z_0,\omega) \quad \frac{\delta \sigma(\omega)}{\delta c(r_0)} = 2 \frac{\omega^2}{c_s^2(r_0)} \sigma_0(\omega) G(r,z_0,\omega) G(z_0,z,\omega)
\]

The variational derivative of the “instantaneous time” \( T \) defined at (11) is:

\[
\frac{\delta}{\delta c(r_0)} \text{Im} \left\{ \frac{1}{\sigma(\omega)} \frac{\text{d}\sigma}{\text{d}\omega_{\text{lo}}} \right\} = \text{Im} \left\{ \frac{1}{\sigma(\omega)} \frac{\text{d} \sigma(\omega)}{\text{d} \omega_{\text{lo}}} \right\} - \frac{1}{\sigma^2(\omega)} \frac{\text{d} \sigma(\omega)}{\text{d} \omega_{\text{lo}}} \frac{\delta \sigma(\omega)}{\delta c(r_0)}
\]

Combined and applied over the background of a uniform unbounded medium (Green function defined by eq. (5)), these equations finally give:

\[
\frac{\delta T}{\delta c(r_0)} = -\frac{1}{2\pi} \left| \frac{r-s}{r-r_0} \right| \left| \frac{r_0-s}{r_0-r} \right| \frac{1}{c_s^2} \left[ \omega_0^2 (\tau_0 - \tau) \cos(\omega(\tau_0 - \tau)) + 2\omega \sin(\omega(\tau_0 - \tau)) \right]
\]
5. DISCUSSION & CONCLUSION

The Time Sensitivity Kernels were intensively studied in the fields of underwater acoustics ([2]) and geophysics ([5], [6], [7]), most generally as a measure of inversibility in a context of tomography. As far as we know, the analysis was almost always conducted using perturbative expansions of the field scattered by sound speed variations (Born or Rytov approximations); these approaches provide explicit expressions of the phase, which when divided by the frequency yield the travel time as an integral linear functional of sound-speed fluctuations. A strange feature of the resulting TSK is the famous “doughnut banana” paradox ([5]): in 3D, the TSK is zero just along the ray-path connecting source and receiver, and features a maximum weight in a tube surrounding this ray. Recently, authors attributed this paradox to Born or Rytov approximations as only first order terms for travel time, to be “corrected” by better definitions of travel time, on the basis of an apparently erroneous analysis of the “instantaneous time” investigated in our section 4 ([6]). We have derived non-perturbative expressions of the TSK using three different definitions of travel times, and different kinds of radiated waveforms; the resulting expressions (9), (10) and (12) of TSK depend on the characteristics of the waveform: mean frequency, bandwidth, pulse length. When the point \( r_0 \), where the sound-speed is perturbed, is approaching the ray connecting source s and receiver r, the term \( \tau_0 - \tau \) tends toward 0. One may check that for \( \tau_0 - \tau = 0 \), all three expressions of the TSK are equal to 0. Our conclusion is that the “doughnut banana” paradox appears for all three definitions of time delay. We have otherwise extended this analysis to stratified environments, using modal expressions of the Green function, with each their phase and group velocities. Later work should have to extend our approach for investigating two problems apparently neglected in the literature about TSK: the effects of noise on sensitivity (maybe secondary in geophysics, but crucial for the classically low signal-to-noise ratios encountered in underwater acoustics), and the distortions of the signals due to channel dispersivity.

REFERENCES

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Organizers: Michael Ainslie, Charles Holland, Dale Ellis and Kevin Heaney
OVERVIEW OF THE REVERBERATION COMPONENT OF TREX13

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Abstract: In the spring of 2013, a shallow water reverberation experiment was conducted to measure contemporaneous acoustic and sufficient environmental data so detailed model/data comparisons could be achieved and important environmental factors could be identified for different applications. The Target and Reverberation Experiment (TREX13) was sponsored by the US Office of Naval Research (ONR) and the Strategic Environmental Research and Development Program (SERDP). It was conducted from April to June of 2013 off the coast of Panama City Beach, Florida, in collaboration with multiple institutions and involving three research vessels: The R/V Sharp, R/V Walton Smith, and the Canadian Force Auxiliary Vessel (CFAV) Quest. From a SONAR viewpoint, reverberation consists of two-way propagation and a single backscatter. Therefore, reverberation, transmission loss, and bottom backscatter were repeatedly measured over a time period of several weeks in the frequency band of 2-10 kHz, along with extensive environmental measurements. To reduce the area over which environmental measurements were needed, the reverberation was measured using a horizontal line array mounted on the seafloor in 19 m of water. The reverberation, transmission loss, and bottom backscatter were measured along a single beam of the array out to a distance of 7 km. Discussed will be planning and execution of the field experiments, strategies and steps for data analysis, and modeling efforts.

Keywords: TREX13, Reverberation, Transmission Loss
1. INTRODUCTION

The detection of underwater targets in shallow water is in many cases reverberation limited. This has led to a plethora of reverberation models that have increased in complexity over time as computational capabilities have improved. Despite the availability of these modeling tools, the relative importance of the various environmental factors that affect reverberation is not well understood. This difficulty stems from the complexity of the reverberation problem, apparent in the sonar equation,

\[ RL = SL - 2 \times TL + SS, \]

where \( RL \) is the reverberation level, \( SL \) is the source level, \( TL \) is the transmission level, and \( SS \) is the integrated scattering strength. For example, in this single scattering approximation, while the backscatter from the sea surface roughness may be weak compared to the seafloor roughness, the surface's affect on transmission loss can still be important for the overall reverberation level [1].

This lack of understanding continues despite a number of careful shallow water reverberation measurements made in different environments and conditions. For naval relevance, these measurements have been made in depths from 50 to 200 m and out to ranges of 100 km [2]. Even using a directional source or receiver, a complete understanding of the environment requires measurements of the surface, water column, bottom, and sub-bottom over a very large area. The scale of this measurement area either means that the environmental characterization is incomplete or that it cannot be performed at the time of the acoustic measurements.

The goal of the Target and Reverberation Experiment (TREX13) was to overcome these difficulties by bringing together a number of acoustic and environmental measurement systems in a very shallow water environment. The TREX13 site, off of Panama City Beach, Florida, is approximately 19 m deep with a seafloor that is predominately sand. While mid-frequency naval sonars do not typically operate in this water depth, a transmitted signal can interact many times with the sea surface and seafloor while traveling only a relatively short range. This is not a scaled experiment where decrease in depth would necessitate a proportional increase in frequency. Instead the physics affecting mid-frequency transmission loss and scattering is the same as in an operational environment but is interrogated over a smaller, shallower area that can be characterized relatively quickly with measurements supported by divers.

2. ACOUSTIC MEASUREMENTS

To measure reverberation due to sound propagation over a manageable area, the Four Octave Array (FORA) was used in collaboration with the Applied Research Laboratory at Penn State (ARL-Penn State) [3]. This research array, built to support ONR-funded basic research, has a triplet sub-array cut for 3750 Hz. This portion produces a cardioid beam pattern that allows for port/starboard discrimination and was used during TREX13. The R/V Sharp was placed in a four-point moor at the experiment site and the FORA was deployed on a frame below ship that held the array horizontally 2.1 m above the seafloor. An ITC-2015 source was mounted on the seafloor behind the FORA and transmitted a suite of pulses from 2-4 kHz.
The reverberation measured by the FORA was beamformed along the two experiment tracks shown in Figure 1: the main track and the “clutter” track. The primary goal was to examine the physics affecting reverberation and hence a majority of the acoustic and environmental measurements took place along the main track. This track started at the location of the R/V Sharp and extended parallel to shore along the isobar running toward the southeast for 7 km.

![Fig. 1: Map of TREX13 showing the distribution of the long term acoustic sources and receivers. The location of the experiment is shown in the inset. The bathymetric data collected along the two main tracks of the experiment is shown with a linear color scale that goes from 12 m below the sea surface (red) to 21 m (dark blue).](image)

While the FORA was collecting data, two vertical line arrays (VLAs), deployed by the Scripps Marine Physical Lab (MPL) from the R/V Walton Smith, were recording the transmitted signal along the main track at ranges of 2.4 km and 4.2 km. These arrays consisted of 32 elements covering the water column from 5 m to 12.2 m above the seafloor with a spacing of 0.2 m. The VLAs recorded continuously throughout most of the experiment. In addition to measuring the reverberation transmissions, the VLAs were also used in several towed source experiments conducted by MPL to measure bottom loss. Towards the end of the experiment, a third VLA was deployed 500 m from the FORA location to serve as a quasi-monostatic vertical aperture.

Transmission loss data were also collected in conjunction with Defence Research and Development Canada (DRDC) using a horizontal receive array towed by the CFAV Quest. This receive array was part of a larger echo repeater system that provided a target for both the reverberation as well as for continuous active source measurements [4,5]. The towed receiver, VLAs, and autonomous recorders provide TL data along the main track as a function of both range and time capturing any effects of the seafloor and sub-bottom as well changes to the sea surface roughness due to weather.

In addition to the VLAs, a benthic tower was deployed by APL-UW on the main track, 5 km from the FORA. A 4-element horizontal line array (HLA), a 7-element VLA, a vector sensor, and a DSG-Ocean Acoustic Datalogger were mounted on the tower. This
tower was deployed for roughly a week in the middle of the experiment and received transmitted pulses from the ITC–2015 that were also received by the VLAs and the FORA. Several towed source measurements were also made from the R/V Smith that utilized the benthic tower arrays to examine vertical and horizontal coherence [6-8]. The acoustic datalogger recorded continuously while the tower was deployed. A second datalogger was also deployed for the length of the experiment, mounted on the seafloor on the main reverberation track at 6 km.

Scattering strength along the main reverberation track was measured from a drifting dive boat using an ITC–1007 as an omnidirectional source and an ITC–1032 as a receiver. Both transducers were mounted on a bracket and suspended approximately 2 m above the seafloor. The source transmitted CW pulses from 2 to 10 kHz and the receiver recorded the field scattered from a patch of the seafloor with a 17 m radius and covering angles from 9 to 90 degrees.

3. ENVIRONMENTAL CHARACTERIZATION

The use of the FORA limited the environmental characterization needed to test models of reverberation to the area defined by the beam pattern along the main reverberation track that had a nominal horizontal width of 2.4 degrees. To further reduce the characterization burden, the primary area for data/model comparison was limited to a range of 5 km (the location of the benthic tower).

Throughout the deployment of the FORA, two systems were monitoring the sea surface conditions. The first was a wave rider buoy deployed to the south of the R/V Sharp. This measured the directional wave spectrum as a function of time during the experiment. The second system was a camera mounted on the mast of the R/V Sharp. Deployed by Scripps, this system captured images of the sea surface to monitor the formation of white caps, an indicator of bubble plume formation due to plunging waves. In addition to these systems, meteorological data were collected by each ship throughout the experiment. A second wave rider buoy was also deployed at a 5 km range along the main track while the benthic tower was deployed.

Additional oceanographic measurements included daily CTD casts from each of the research vessels and temperature measurements by thermistor chains attached to the Scripps VLAs. These measurements show that the sound speed profile was approximately isovelocity throughout the experiment. Measurements of the current were collected on a second benthic tower deployed to the south of the R/V Sharp location for the duration of the experiment. This system was deployed by NRL-Stennis to measure sediment transport and munitions mobility at the site as part of a separate SERDP-funded effort [9-11]. In addition, a pencil beam sonar on this second benthic tower also measured sea surface wave heights throughout the experiment.

A significant effort was made to characterize the seafloor at the experiment site. A multibeam sonar survey was conducted by the company “10dBx,” both before and after the experiment to provide high-resolution bathymetric data and to monitor any changes that might have occurred at the site during the experiment due to significant weather events [12]. In addition to bathymetry, the multibeam data were also processed for backscattering strength at 400 kHz. Sediment samples were able to confirm that the high backscatter areas at 400 kHz were composed of sand with significant shell content and the low backscatter areas were predominately mud.

Bottom roughness was measured at a number of locations along the track using two laser line-scanning systems. The first system was deployed by the Applied Research Laboratory, University of Texas (ARL:UT) and is mounted on a remotely operated vehicle
to collect roughness data over a large area of the seafloor [13]. This data were collected along with bottom loss measurements conducted by both ARL:UT and the Acoustic Research Laboratory of the National University of Singapore (ARL-NUS) [14]. The second system was deployed by APL-UW and is mounted on a frame to measure high-resolution roughness over a smaller portion of the seafloor. This system also collected porosity data in the top 20 cm of the seafloor using a conductivity probe.

![Image](image.png)

**Fig. 2: Example of reverberation data measured at the TREX13 site during calm conditions (Provided by Jie Yang APL-UW).**

In addition to the ARL:UT bottom loss measurements, reflectivity was also measured by ARL-Penn State and DRDC. Both of these measurements can indirectly provide sound speed in the sediment. Direct measurements of the sound speed were made using the APL-UW Sediment Acoustic Measurement System (SAMS) that measures the time-of-flight of signals transmitted from the seafloor to a receiver 3 m below the sediment interface.

Sub-bottom profiler data were collected by both the Institute for Geophysics at the University of Texas (UTIG) and DRDC. These data sets provide information about the sediment structure beyond 50 cm below the sediment interface. To characterize the upper portion of the sediment, the Naval Surface Warfare Center Panama City Division (NSWC-PCD) deployed the Buried Object Scanning Sonar (BOSS) along the main track [15]. This system is typically used to image mines and ordinance buried in the seafloor, but was deployed here to look for heterogeneities in the seafloor.

Finally, in an effort to characterize the fish activity at the experiment site, UW and the Wood’s Hole Oceanographic Institute (WHOI) conducted several surveys of the site using an echo sounder to map the fish distribution as a function of time and location.

4. **DATA ANALYSIS**

Preliminary analysis of the acoustic data collected during this experiment has focused on understanding the role played by each term of equation (1) in reverberation under different conditions. To facilitate analysis, the reverberation data are being sorted and divided into the categories that emphasize a particular environmental impact: bottom reverberation under calm sea surface conditions (baseline dataset), reverberation under different wind conditions, reverberation dominated by biologics, and reverberation with
both targets and clutter present. An example of the baseline reverberation data is shown in Fig. 2.

![Graph showing reverberation data](image)

**Fig. 3: Reverberation measured during TREX13 when there was a high sea state (red curve) and a low sea state (blue curve).**

While the analysis is still in its early stages, some interesting results have already emerged. While the water depth changes only modestly along the main track, the reverberation level fluctuates on the order of 10 dB. These fluctuations are closely correlated to topographic changes, however the modest topographic variation is easily ruled out as the mechanism for the fluctuation. As seen in Fig. 1, the bottom has a ridge and swale topography with significant shell content on the southeast faces of the ridges and mud in the swales. The 400 kHz backscatter is strongest on these southeast faces while the 3 kHz backscatter is strongest in the mud-dominated swales. This strongly suggests that subbottom scattering at the mud locations is the cause of the reverberation fluctuations.

One of the main goals of the experiment was to collect reverberation under the influence of different sea surface conditions. While scattering from the seafloor dominates the backscatter in reverberation, models have shown that without taking into account both the coherent loss due to sea surface forward scatter and the incoherent scattered field, predictions of reverberation can be off by as much as 10 dB under a 15 knot wind [1]. Preliminary TREX13 data analysis from two different days, one calm, the other with high wind, show different reverberation levels qualitatively consistent with these predictions (Fig. 3). Systematic analysis of data under a wide range of wind and swell conditions will be performed to establish the importance of forward surface scattering.

**REFERENCES**


CORRELATION OF REVERBERATION WITH BOTTOM SAND WAVES ALONG THE TREX REVERBERATION TRACK

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Abstract: The Target and Reverberation EXperiment (TREX13) took place in the Gulf of Mexico just off the coast of Panama City, Florida. The reverberation experiments were conducted, weather permitting, between 22 April and 16 May. Both source and receiver, ITC 2015 and the Five Octave Research Array (FORA), were deployed up to 50 m apart, fixed in location, and 1.2 and 2.1 m above the seabed respectively. Of particular interest here are various pulses between 1800 Hz and 3600 Hz, with the latter frequency being near the design frequency of the FORA triplet array. During TREX13, reverberation data were taken during all hours of the day and night, allowing study of reverberation variation over time and sea surface conditions. In addition to the time and weather dependence, the full 360° directional dependence of reverberation level (RL) could be determined using the FORA triplet array. The focus of the experiment was on RL returning from a track of relatively uniform water depth of 20 m, extending about 10 km to the southeast of the source and receiver. The most interesting observation was the effect of the bottom sand waves or dunes, roughly 1 m peak to trough and spaced about 300 m apart, on RL. This correlation between the two had been noted in the pre-TREX experiment in 2012 but not analysed in detail. Predictions from an adiabatic normal mode reverberation model were used to compare with measurements, using the detailed bathymetry, but otherwise with inputs independent of range. As expected, the model predicts a peak in the reverberation at the peak of the sand dunes. However, counterintuitively, the peaks from the data are, more often than not, anti-correlated with the peaks of the bathymetry, i.e. high RL correlated with the troughs of the sand dunes. Clearly, some mechanisms other than depth effects are responsible for the changes in RL. Extensive bathymetric and bottom measurements have been made along this track; these are being investigated by other researchers to facilitate understanding of the reverberation mechanisms.

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Keywords: reverberation, clutter, range-dependent modelling, normal modes, TREX Experiment, sand dunes, Gulf of Mexico, towed array, cardioid beam patterns.
1. INTRODUCTION

TREX was a series of Target and Reverberation Experiments sponsored by the US Office of Naval Research (ONR) in the Gulf of Mexico just off Panama City, Florida, US. Their unique feature was a fixed source and fixed receiver deployed in about 20 m of water, with an extensive set of complementary environmental measurements to facilitate the understanding of the underlying reverberation and clutter mechanisms, and to support quantitative modelling. The experiments were organized by Applied Physics Laboratory, University of Washington (APL/UW).

An initial acoustic experiment was conducted in April 2012, with the fixed source and a horizontal array deployed from a moored vessel. The main experiment was conducted in April and May 2013, with the source and receiver similarly deployed from RV Sharp. In addition, Defence Research & Development Canada (DRDC) participated with their research vessel CFAV Quest, towing an echo repeater for target echo and transmission loss measurements. Other equipments were deployed, including the Scripps vertical line arrays for transmission loss measurements, and a DRDC passive acoustic target (PAT) for echo measurements from a fixed location.

Our main focus was reverberation experiments which were conducted, weather permitting, daytime and nighttime, between 22 April and 16 May 2013. Particular attention was directed to reverberation returning from a track of relatively uniform water depth, extending about 10 km to the southeast of the source and receiver. An overview of the experiment is summarized by Tang and Hefner [1, 2], and the target echo measurements are discussed by Murphy et al. [3]. Preston [4] has presented an overview of the some of the data.

A most interesting observation was the effect of the bottom sand waves or dunes, roughly 1 m peak to trough and spaced about 300 m apart, on the reverberation. A correlation between the two had been noted earlier [5] in the 2012 data. This paper explores this effect in more detail. The key observation is that the peaks in the reverberation seem to be correlated with the troughs of the sand dunes, rather than the crests of the sand dunes as one would expect.

In this paper we explore some of the data and its anti-correlation with the bathymetry, with the help of some model predictions. The explanation is likely related to the bottom or sub-bottom effects. Our purpose is not to explain the results, but rather to focus on what needs to be investigated further in order to determine the underlying reverberation mechanisms. Extensive bathymetric and bottom measurements have been made along this track; these are being investigated by other researchers to facilitate understanding of the reverberation mechanisms.

2. EXPERIMENT LAYOUT

Figure 1 shows the bathymetry and some clutter objects near the experiment site. RV Sharp was moored near 30°03.59′N 85°40.86′W in 18.3 m of water. The source was an ITC 2015 deployed about 1.2 m above the seabed on a fixed tower. The receiver was the triplet section of the Five Octave Research Array FORA [6], deployed horizontally as a fixed receiver about 2.1 m above the seabed. They were deployed in several geometries during the experiment; one of which is included in Fig. 1. Table 1 gives our best estimates of the source and receiver locations.

Other equipments were deployed, among them the DRDC Passive Acoustic Target (PAT), a 15 m vertical air filled hose, at location 30.04897°N 85.65422°W about 2.8 km from Sharp, and the Scripps vertical line arrays (VLAs) at ranges of 0.5 km, 2.4 km and 4.2 km from Sharp along the main reverberation track on bearing 129°. For part of the time the DRDC research vessel CFAV Quest towed an echo repeater for transmission loss and target echo [3] experiments. Other environmental measurements were made as well.

Each element of FORA’s triplet section has three hydrophones which can be used to form broadside cardioid beams [7]. In the 2012 and 2013 experiments 48 elements of the FORA triplet array were used to form beams in the full 360° azimuth; this left-right discrimination was essential for this experiment. The element spacing in FORA is 0.2 m, which for a sound speed of 1520 m/s implies a spatial aliasing frequency of 3800 Hz. The data were sampled at 12.5 kHz and processed.
in different ways. The Applied Research Laboratory (ARL) and DRDC used Preston’s time-domain beamformer which formed 157 beams spaced equally in cosine of the steering direction, clockwise from forward endfire; the elements were Hann weighted. APL used a frequency-domain beamformer; the beam angles could be chosen arbitrarily, often at 1° increments; the elements were uniformly weighted, which gives narrower beams but higher side lobes.

Various pulses in the 1800–3600 Hz band were sent from the ITC 2015 source. The APL pulses were typically 1 s in length, transmitted with constant voltage across the band, with 10% Tukey shading at each end. The DRDC pulses were typically 0.5 s in length, with 5% Tukey shading at each end, and voltage adjusted across the band to compensate for the voltage response of the projector to produce a constant source level as a function of frequency.

The results were matched filtered in different ways. The APL processing uses a Gaussian band pass filter, and then does the correlation; the results are normalized to produce output in energy. Preston uses a complex bandshifting procedure, and his matched filter reverberation and noise are reduced by approximately the time-bandwidth product [8]. DRDC used a processing chain very similar to Preston’s, but the results were not calibrated. The matched filtering is a correlation, but for modelling later we use an energy model, so it is important to understand how the matched filter levels correspond. This work is still in progress, but we think that the matched filter levels from the APL processing align closely with the ping energy levels, so have used that data for the quantitative comparison.

3. MEASUREMENTS

Reverberation data were taken during all hours of the day, allowing study of reverberation variation over time and sea surface conditions. In addition to the time and weather dependence, the directional dependence of the reverberation could be determined using the FORA triplet array.

Figure 2 shows a polar plot of the normalized, averaged, reverberation from Run 44. In this case FORA was deployed at bearing 219°, so broadside beam 118 points down the reverberation track, and beam 40 in the opposite direction. Note the north-south striations in the data to the west and southeast of the array.
Table 1: Best estimates of source-receiver geometry and locations. In this paper we use the upper locations from the sea trial; the locations in parentheses are from follow-on work at APL.

<table>
<thead>
<tr>
<th>Date</th>
<th>Runs</th>
<th>Julian day</th>
<th>ITC location Lat</th>
<th>Long</th>
<th>Approx. src.-rec. geometry</th>
<th>FORA Nominal heading</th>
<th>Estimated orientation</th>
<th>Module centre Lat</th>
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<td>Runs 11–32</td>
<td>JD113–116</td>
<td>30.05977 –85.68065</td>
<td></td>
<td>source 3 m NW of FORA centre</td>
<td>219</td>
<td>217.5</td>
<td>30.05976 –85.68063</td>
<td></td>
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<tr>
<td>Apr 26–May 2</td>
<td>Runs 33–53</td>
<td>JD116–122</td>
<td>30.05976 –85.68117</td>
<td></td>
<td>source 50 m W of FORA</td>
<td>219</td>
<td>217.5</td>
<td>30.05976 –85.68063</td>
<td></td>
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<tr>
<td>May 7</td>
<td>Runs 58–62</td>
<td>JD127</td>
<td>30.05976 –85.68100</td>
<td></td>
<td>source 62.5 m N of FORA centre</td>
<td>333</td>
<td>333</td>
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</tr>
<tr>
<td>May 8–14</td>
<td>Runs 63–131</td>
<td>JD128–136</td>
<td>30.05976 –85.68100</td>
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<td>source 62.5 m N of FORA centre</td>
<td>358</td>
<td>353</td>
<td>30.05920 –85.68105</td>
<td></td>
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</table>

Fig. 2: (Left) Polar plot of reverberation for 2600–2800 Hz LFM pulse from Run 44 with DRDC processing, averaged and normalized by Mathieu Colin from TNO; the black circle marks 5 s. (Right) Multibeam survey over a 1.3 by 7 km track, showing sand dunes 1–2 m in height, several hundred meters apart; the source/receiver were located midway on the NW side of the box.

Sand dunes, and their migration, are known in this area [9], so a number of bathymetric surveys were conducted. Figure 2 includes one conducted by de Moustier and colleagues before the experiment. A correlation between the reverberation and the sand dune had been noticed previously [5] in the 2012 data, but not averaged or carefully compared with the bathymetry.

In the 2013 experiments it was observed that the higher reverberation levels corresponded more often with the troughs of the sand dunes, rather than the peaks. Figure 3 shows a section of a polar plot of the reverberation fluctuations from Run 82 at short ranges, with some of the bathymetric contours overlaid. Other than at very short times, the low scattering seems to be coming from the peaks of the dunes (white contours), and the high scattering from the troughs (dark contours). This seems somewhat counterintuitive.

4. INTERPRETATION USING MODELLING

We now look at the data in more detail using a normal-mode reverberation model. Here the intention is not to provide accurate predictions, but to provide insight into understanding the data.
The Clutter Model can calculate reverberation and target echo on a towed array in bistatic geometry for a range-dependent environment [10, 11, 12]. It uses adiabatic normal modes for propagation, ray-mode analogies for scattering, and calculates the time series of reverberation, echo, and clutter, for all beams over an area. Calculations have been presented in previous papers as part of planning for the TREX experiments [12, 13, 5]. The data in Fig. 3 are averaged reverberation for a number of the 1800–2700 Hz LFM pulses. The model is a flat-bottom Clutter Model prediction at 2250 Hz.

For more detailed calculations we use a simpler model called R2D3 [14], which performs calculations for monostatic geometry along a single radial. We first try to see if the matched filter reverberation levels look quantitatively reasonable by comparing with a model prediction.

The key model inputs for the calculation were: isovelocity water column with sound speed 1530 m/s, and Thorp volume absorption; bottom halfspace with sound speed 1680 m/s, relative density 2.08, and attenuation 1.0 dB/m-kHz; bottom scattering with Lambert coefficient of $-27$ dB. Two bottom profiles were used: (1) flat bathymetry, with water depth of 18.5 m; and (2) an approximation to the bathymetry along the reverberation track at 10 m spacings.

Figure 4 (left) shows a comparison of the range-dependent R2D3 model with some Run 17 data, a 10-ping average in the 3400–3500 Hz band. The model was run at 3450 Hz with energy source level ESL of 198.4 dB; the omnidirectional results were reduced by the effective reverberation response [15] of 19.7 dB. The predicted reverberation is about the correct level, although a slower drop off at short ranges and faster drop off at long ranges would be better. (A few variations on the input parameters were tried; e.g., doubling the bottom attenuation, and reducing the sound speed to 1600 m/s. These did not seem to improve the model-data agreement). These model-data comparisons are not meant to be definitive in any way; rather, they are a rapid environmental assessment tool [15] to determine where we should concentrate for more data analysis and improved environmental inputs. Similar graphs (not shown) were obtained for the 1900–2000 Hz LFM and 2700–2800 Hz LFM, using ESLs of 195.1 and 196.8 dB, and effective reverberation response of 17.2 and 18.7 dB respectively. This accounts for most of the frequency dependence in the data. The bottom reflectivity for a half space is frequency independent, and the Lambert coefficient was frequency-independent, so it suggests the scattering is not strongly frequency dependent.

Note that at short range the data shows much higher variations than the model prediction. First
we look at the model predictions. Figure 4 (right) shows the the flat-bottom model prediction subtracted from the range-dependent model prediction; also shown is input bathymetric profile (offset by 20 m). As the water gets shallower, the reverberation increases. This is what we would expect. (The model takes into consideration the bottom slope – however it is negligible, the order of 0.5 m in 150 m, or about 0.2°.) The data, as we have seen, shows the opposite behaviour with respect to the bathymetry, and we will look at it in more detail.

Figure 5 compares the reverberation variations – data minus the flat-bottom model prediction – (black) vs the bathymetric profile (red) and the slope (green) of the bathymetry. Note that the depth (red) is positive here, so that the peaks in the reverberation (black) roughly correspond to the dips in the bathymetry (counter-intuitive). The upper graph is from the first deployment of the source and receiver – FORA near the start point of the reverberation track. The data are from Run 17 (10-ping average of the 3400–3500 LFM processed at APL); the model is as described earlier for Fig. 4. The lower graph is from the third deployment of FORA – about 50 m west of the start of the reverberation track. The data are from Run 82, the 1900–2700 Hz LFM processed by the DRDC system (uncalibrated). The model calculation was at 2250 Hz, with similar environmental inputs, and with the mean level adjusted to approximate the correlated data.

Though not perfect, there is definitely a correlation between the peaks of the measured reverberation and the troughs in the bathymetry. Fluctuations, and side lobes will likely have some effect. The correlation with the slope is not as good. The explanation for the reverberation variations is likely related to the seabed or sub-bottom effects. This points to the need for additional measurements, especially in the trough regions, since that is where the measured reverberation levels are higher than the canonical −27 dB Lambert coefficient which approximates the bottom envelope of the reverberation at short ranges in Fig. 4.

5. SUMMARY

The TREX13 experiment was extremely successful. High quality data were collected, during both daytime and nighttime, for almost a month. The source levels were kept low for environmental reasons, but the directional nature of the FORA array coupled with narrow band LFMs allowed signal to ambient noise out to 5 s and in some cases to 10 s.

The area had been chosen to be flat and uniform in order to facilitate modelling. However, one of the striking features was the correlation of the reverberation features with the sand waves (or dunes) in the area. Although only about 1 m peak-to-trough over several hundred meters range, they resulted in reverberation fluctuations on the order of 10 dB.
Fig. 5: Correlation between the averaged reverberation along the main track, the bathymetry, and the slope of the bathymetry for Run 17 (upper) and Run 82 (lower). The bathymetric profile is from the acoustic centre of the FORA array toward the 7 km point of the reverberation track, and offset by 19 m for display. The slope (upward being positive) is increased by a factor of 100. The model-data differences in dB are reduced by a factor of 10; i.e., units of Bels!

Some model-data comparisons were made — not to be definitive in any way, but rather as a rapid environmental assessment tool to determine where the research team should concentrate for more data analysis and improved environmental measurements.

A key observation is that the peaks in the reverberation seem to be correlated with the troughs of the sand dunes, rather than the crests of the sand dunes as one would expect. The simple model predicts the opposite behaviour, and smaller peaks. So this means there is some interesting physics to be explored, which is beyond the scope of this paper.

The explanation is likely related to the bottom or sub-bottom effects. Extensive bottom, sub-bottom, and other environmental measurements have been made along this track; these are being investigated by other researchers to facilitate understanding of the reverberation mechanisms, which was indeed the purpose of the TREX experiment. We will work toward improved models which include this new understanding.

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Much of the work was supported by the US Office of Naval Research, Code 322 OA. DJ Tang and Kevin Williams from APL/UW (Applied Physics Laboratory, University of Washington) directed the experiments. Paul Hines from DRDC Atlantic was Chief Scientist on the DRDC research vessel CFAV Quest. Akoostix Inc. processed data for DRDC. Mathieu Colin from TNO (Netherlands) spent a week aboard Quest and was very helpful in getting a quick look at the data. Support from the crew and staff on CFAV Quest and RV Sharp are most appreciated.

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REFERENCES


A FAST ALGORITHM FOR THE COMPUTATION OF INCOHERENT PROPAGATION LOSS FOR VARIABLE WATER DEPTH: A VALIDATION STUDY

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Abstract: Accurate and fast estimation of propagation loss (PL) is needed for simulations of sonar or acoustic communication performance, and for environmental risk assessment. Accurate calculation of PL in range dependent and lossy waveguides can require computationally expensive wave theory techniques. In this work, a new propagation model is introduced for a range-dependent isovelocity waveguide. The proposed model is based on a combination of Weston’s average intensity approach and adiabatic normal mode theory. Thus, mode effects arising at both low frequencies and long ranges are automatically considered. Using this approach, the depth and frequency dependence of PL can be calculated analytically in terms of the Faddeeva function without tracing rays or calculating normal mode eigenvalues (except low order modes). For the validation, PL results are compared with KrakenC and Bellhop by using test cases from Weston Memorial Workshop 2010. Comparison with results computed independently using a full wave model demonstrate the accuracy of the model applied to WMW 2010. Thus, the proposed solution can be an alternative solution for large scale problems such as broadband calculations in sound mapping without requiring long computational times.

Keywords: Normal modes, flux theory
1. INTRODUCTION

Propagation loss (PL) can be calculated by different methods such as normal mode, parabolic equation, ray tracing and energy flux theory. All of these methods can be based on the different approaches for the range-dependent problems. The performance of these method were compared [Sertlek&Ainslie,2013]. Normal mode solutions can provide one of the most stable and accurate estimate for the propagation loss. The depth dependent properties of incoherent PL in the vicinity of sea surface seabed, source depth and complementary source depth can be modelled with mode theory. However, the calculation of eigenvalues and the use of stair-step approximation for the bathymetry can lead to long computational times for the high frequencies. On the other hand, Weston’s average intensity approach can provide a fast analytical solution for the range-dependent problems which is based on the effective depth concept. The weakness of this approach can be seen at low frequencies and long ranges where only a few modes propagates. The dependence of PL including the effect of surface decoupling is not considered in the average intensity approach[8]. In this paper, a hybrid algorithm SOPRANO (Source and Propagation Algorithms for Noise Assessment), is introduced for shallow water propagation. SOPRANO combines the speed of flux theory and accuracy of mode theory including the depth dependent properties of PL. The obtained results are compared with KRAKENC by using Weston Memorial Workshop 2010 scenarios[1-3]. The detailed agreement is obtained between these PL results.

2. WESTON MEMORIAL 2010 TEST CASES

Test problems from the 2010 Weston Memorial workshop [1,2] are considered. These are based on the test problems for the 2006 and 2008 Reverberation Modelling Workshops at Un.Texas at Austin [3]. For Scenario A2.I , the water depth (h) is 100 m, source depth (z_s) is 30 m, sound speed in water (c_1) and sediment are respectively are 1500 m/s and 1700 m/s. The sediment-seawater density ratio is 2 and resulting reflection loss gradient (η) is approximately 0.2738 Np/rad. Scenarios A2.IV has a problem with upslope bathmetry. In this paper, they are called as “Case 1” and “Case 4” respectively for range independent and range dependent cases[5,7]. It has 100 m water depth up to range 5 km. Then, it has upslope from 5 km to 7 km up to water depth 30 m.

![Figure 1. A range dependent test case from WMW 2010 (Case 4 or Scenario A2.I)](image-url)
3. HYBRID METHOD FOR PROPAGATION LOSS

PL can be described as

$$PL = 10 \log_{10} \left( \frac{F^{-1}}{r_{ref}^2} \right) \text{ dB re } 1 \text{ m}^2$$ (1.1)

Where $F$ is propagation factor and $r_{ref}$ is reference range. Propagation factor can be calculated by Weston’s flux integral as [10]

$$F_{\text{Weston}} = \frac{2}{rH_s} \int_0^{\theta_m} \exp \left( -\eta r H_{eff}^2 \theta_s^2 \right) d\theta_s$$ (1.2)

where $H_{eff}$ is the effective depth given by $H_{eff} = \frac{H_s^2 H_r^2}{r} \int_0^r dR$, $\eta$ the reflection loss gradient, $H_s$ the water depth at source, and $H_r$ is the water depth at receiver. Eq.(1.2) can be solved analytically as

$$F_{\text{Weston}} = \frac{\sqrt{\pi}}{r^{3/2} \eta H_{eff}} \text{erf} \left( \frac{\theta_m}{H_r} \sqrt{\eta r H_{eff}} \right)$$ (1.3)

and erf is the error function, $r$ is the range, $\theta_m = \min \left\{ \frac{H_s \theta_c}{H_s} \right\}$, $\theta_c$ is the critical angle and $\eta$ is the rate of increase with grazing angle of the sediment reflection loss [10]. For long ranges, the error function approaches to 1. Although this equation provides rapid and analytical solution for the range dependent problems, it does not take into account the depth dependence of propagation factor. In order to model to depth dependence, an extra term ($W$) is derived from mode theory, and added to the flux integral as shown by Eq.(1.4). The dependence of PL due to various receiver and source depth combinations can be modelled with this depth-dependent correction term [11],

$$F = \{\text{ModeSum}\} + \frac{2}{r H_s} \int_{\theta_{ass}}^{\theta_m} \left( 1 - W(z_s, z_r, \theta_s) \right) \exp \left( -\eta r \frac{H_{eff}^2}{H_r^2} \theta_s^2 \right) d\theta_s$$ (1.4)
A contribution from lower order modes is also separately added to this integral. This contribution improves the accuracy especially for the low frequencies and long ranges. After these modifications for the flux integral, propagation loss can be estimated accurately as mode theory. The depth dependent properties of PL are also considered. In Section 5, some comparisons will be shown.

4. COMPARISONS

In comparisons, it is sometimes difficult to see the small details when the PL are compared. Instead of the direct comparison of PL, the relative PL can be compared. The relative PL is defined as

\[ \delta \text{PL} = \text{PL} - \text{PL}_{\text{Ref}} \]  

(1.5)

Where PL is the obtained result of the used propagation algorithm, and \( \text{PL}_{\text{Ref}} \) is a reference PL which can be simply obtained for all scenarios. Having a reference PL which is analytical, numerically stable and fast is preferable. Weston’s average intensity method which has all of these properties is a good candidate to use as a reference PL. In these comparisons, the reference PL is calculated from Eq.(1.2) as \( \text{PL}_{\text{Ref}} = \text{PL}_{\text{Weston}} \).

The relative PL versus range (up to 45 km) is plotted for the 10 m receiver depth for CASE 1 and CASE 4. In all test cases, the sound speed in water is 1500 m/s. The sound speed in sediment is 1700 m/s and bottom absorption loss is 0.5 decibels per wavelength \([0.294 \text{ dB/} (\text{m kHz})]\). In figure 2, the relative PL for CASE 1 and CASE 4 is shown at 1 kHz and 10 m receiver depth. Only one mode is used for the mode region contribution.

![Relative PL for range independent (CASE 1) and range dependent (CASE 4) test cases at 1 kHz. The receiver depth is 10 m.](image-url)
For the range independent scenario (CASE 1), very well agreement is obtained between KRAKENC and SOPRANO. The difference between Kraken and SOPRANO is less than 0.0013 dB. The maximum difference for CASE 4 is around 0.3 dB. Solution for CASE 4 can be improved by adding more modes to mode region. In mode theory calculations, the stair-step approximation of bathymetry is used. The length of these stair-steps also effects the accuracy of mode theory. For this comparison, the stair-step length is 20 m for KRAKENC. SOPRANO is based on piecewise-linear approximation of bathymetry.

5. CONCLUSIONS

A new propagation model (SOPRANO) which is based on mode and flux theories is introduced. This model can generate quite fast results for the range dependent problems without numerical calculation of complex eigenvalues or tracing rays. The lower order modes are estimated analytically and added to flux integral. The detailed agreement between KrakenC is obtained both range and depth dependence of PL. The proposed method can be used for the large scale acoustic propagation problems such as sound mapping.

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A SURVEY AND A GENERAL EVALUATION OF CLASSICAL SURFACE LOSS MODELS FOR LOW FREQUENCY SONAR PERFORMANCE EVALUATION

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Abstract: We review and evaluate the most classical formulas for sea surface acoustic loss, widely used in the field of underwater acoustics. We classify and analyse the different existing algorithms, for frequencies spanning from a few 100 Hz up to some 25 kHz:

1) Theoretical formulas relying on the effective loss due to surface roughness only, like the Marsh, Schulkin and Kneale formula (1962), with corrections by Kuo (1988). Effects due to other phenomena like loss due to near surface air bubbles are by principle discarded in such algorithms.

2) The strange case of the formulas referred to Beckmann and Spizzichino, that have nothing to do with these authors, is considered; we follow and clarify the story of these formulas down to their very first origin in recently declassified documents: an empirical formula established in 1962 by Marsh and Schulkin, from a set of experimental data with no information concerning dependence on incident angle.

3) Empirical formula of loss in configurations where the sound field is confined near the surface: surface channels associated with mixing layers in deep waters (AMOS experiments in the 1940’s – 1950’s; Saxon & Baker measurements, 1955, etc.); shallow sea channels with hard sandy or gravelly bottom (Weston and Ching data of the 1960’s and formulas, 1989).
Recommendations for the choice of algorithm and their applicability domain are given.

Keywords: sonar performance, sea surface, surface loss.
1. **SURFACE LOSS IN SONAR PERFORMANCE ASSESSMENT**

Two main phenomena affect the reflection of sound on the sea surface: first, the scattering from the moving rough sea surface, which spatially distorts the wave-fronts and temporally and spectrally spread the transmitted waveforms; second, the absorption of acoustic energy due to population of air micro-bubbles. Both mechanisms make surface reflected sound fields depart from perfectly specular images reflected on a plane pressure release upper boundary; this deviation may be, as a first approximation, described by a reflection coefficient $R$, or after applying a logarithm, a surface loss:

$$\text{Loss}_{\text{dB}} = -20 \log_{10} R = -10 \log_{10} R^2$$

The surface reflection loss, generally expressed as a decrease of acoustic intensity in dB per surface bounce, is of a crucial importance in sonar performance assessment, particularly in ocean configurations featuring upward-refracting sound-speed profile. Such configurations include surface channels associated with mixing layer, and arctic or winter isothermal waters. These situations induce multiple successive reflections of rays on the surface, so that surface loss applies cumulatively, several times; accurate modelling of the single reflection loss seems quite primordial for evaluating probability detections. In this paper, we list and evaluate the most classical models of surface loss.

2. **SMALL-PERTURBATION MODELS OF THE SPECULAR PART**

Historically, a first approach to the complex modelling of sea surface reflection was to ignore air bubble effects, and to consider rough surface scattering only; moreover, the “useful” reflected field may be supposed to reduce to its specular coherent part alone, discarding its scattered part as “noise” that would give no sonar gain. Such a modelling may invoke Rayleigh (end of XIX° century):

$$R = \exp\left(-2k^2\sigma^2\sin^2\theta\right)$$

where are involved the standard deviation of wave height $\sigma$, the grazing angle $\theta$ and the acoustic wave-number $k$. More recently, Marsh, Schulkin and Kneale ([1]) have proposed perturbative formulas (low frequency, low grazing angle and low wave-height):

- Marsh, Schulkin & Kneale (1961)
  $$R^2 = 1-0.485\left(f_{kHz}H_{ft}\right)^{1.5}H_{ft}^{0.1}\sin\theta$$

- Marsh, Schulkin & Kneale, corrected by Kuo ([2], 1988)
  $$R^2 = 1-0.165\left(f_{kHz}H_{ft}\right)^{1.5}H_{ft}^{0.1}\sin\theta$$

- Kuo ([2], 1988)
  $$R^2 = 1-1.51610^{-4}f_{Hz}^{1.5}\sigma_m^{1.6}\sin\theta$$

$\sigma_m =$ standard deviation of wave height, in m  
$H_{ft} =$ average peak-trough height, in ft

Small perturbations are conducted up to order $0[\sigma^2]$; only the specular term is considered. Validity: formulas are applicable only as far as Loss < 3 dB, i.e. $H < 8$ ft.kHz (Marsh & Schulkin, [1]). After Kuo ([2]), condition to be assumed: $\sigma/\lambda \sin\theta << 1$. All this converges to an application valid only for Ultra Low Frequencies, where bubble effects can be ignored, and where the specular coherent part is dominant. Typically, for sea-states 2 to 4, in surface channels where typical grazing angles are less than about 2 to 3°, MSchK models can be recommended only below a few 100 Hz.
3. THE SO-CALLED “BECKMANN AND SPIZZICHINO” MODELS

A group of more complex formulas for surface loss, referred to Beckmann and Spizzichino (sometimes wrongly spelled as “Beckmann and Spezzichino”), has widely spread in the world of sonar and almost ranked as a de facto standard, even if sometimes questioned ([3]) or severely criticized. The most intriguing feature of this model is that the invoked authors have nothing to do with the formulas attributed to them. Several versions coexist, taking their origin from US sonar performance programs of the 70’s and 80’s (NISSM II, RAYMODE). Recently declassified documents, readable on Internet, help in clarifying the origin and limitations of Beckmann and Spizzichino (BSp) formulas. They consist in the sum of two terms: 
\[ \text{Loss}_{\text{SL}1} + \text{Loss}_{\text{SL}2} \]. Their characteristics are collected in Table 1.

The first term takes its very origin in a set of at-sea data, collected by Marsh and Schulkin ([4]), probably in surface channel experiments within the 1949-53 AMOS campaign (see Fig.1). The SL1 term consists in a sigmoid analytical fit of these data, which are surface-loss per surface bounce as function of the product frequency \( x \) average trough-crest wave-height product; plots of three versions of this fit, over the original data, are displayed by Fig.2. No dependence on grazing angle is considered, due to very low angles in surface channels (less than 3° for surface channel depths less than 100m).

The second term is a theoretical High-Frequency extension to the first term, including only surface roughness effects. It is derived from the theory of surface scattering strength proposed by Beckmann ([5]), in the VHF, very rough sea surface limit \( (2\pi\sigma/c >> 1) \), under not too grazing incidence (no shadowing). The formulas themselves for SL2 appear nowhere in Beckmann and Spizzichino’s book, often quoted as the origin of the model. Following the analysis of BSp formulas by Lavor ([6]), terms SL2 seem to be approximate expressions of \( 1-B \), where \( B \) is the integral on solid angle, over the full half-space, of the scattering strength \( \Sigma \) evaluated by Beckmann ([5], p.89):

\[
B(\theta) = \int_0^{\frac{\pi}{2}} d\theta_3 \int_0^{\frac{\pi}{2}} d\theta_2 \sin \theta_2 \Sigma \\
\Sigma = \frac{a}{\pi \tan^2 \beta_0} \exp\left(-\frac{1}{2} \frac{\phi}{\tan^2 \beta_0}\right) \frac{F^2}{(\cos \theta_1 + \cos \theta_2)^2}
\]

We use Beckmann’s notations (Fig.3.2, p.18, in [5]) for angular variables \( \theta_1, \theta_2, \theta_3 \). The parameters \( \phi \) and \( F \) involved in this last equation are listed in Table 2, where appear the probably erroneous expressions of Lavor and our understanding of Beckmann’s theory. We successfully compare the different versions of SL2 to numerical evaluations of the two versions of the integral 1-B on Fig.3, confirming Lavor’s interpretation of SL2. Physically, SL2 is the fraction of incident energy that is not scattered, i.e. the specular part; no effect of air micro-bubbles is considered.

As experimental, the analytical fit SL1 deserves consideration, even if limited to low \( fH \) product (\( fH < 100 \text{ ft kHz} \)): for sea-states 2 to 4 and grazing angles less than some 3°, this corresponds to applicability valid up to several kHz, and should work successfully in surface channels or shallow waters configurations. On the contrary, the HF term SL2 has to be handled with care: it is theoretically limited to very high frequency and high grazing angles (because it relies basically on Kirchhoff approximation); it lacks the effects of shadowing under grazing incidence and of attenuation by bubbles; it considers only the specular coherent part of the reflected field.
Fig. 1: The original plot of experimental data used in the Low-Frequency term SL1 of Beckmann & Spizzichino models (p. 48, in Marsh & Schulkin [4], 1962).

\[ SL1 = 20 \log_{10} \left( 0.3 + 0.7 \left(1 + A^2\right) \right) \]

\[ SL2 = 10 \log_{10} \left(1 - B\right) \]

<table>
<thead>
<tr>
<th>Version</th>
<th>Term SL1</th>
<th>Term SL2</th>
</tr>
</thead>
</table>
| NISSM II (1973) | \[ A = \frac{f_{\text{kHz}} H_{\text{fl}}}{40.} \]
                             \[ H_{\text{fl}} = 0.0182 V_{\text{km}}^{5/2} \] | \[ B = \sin \theta_{\text{rad}} + \exp \left(-\frac{1}{4} \sigma \theta_{\text{rad}}^2\right) / \sqrt{\pi \sigma} \]
                             \[ \text{with } \sigma = \frac{500.}{3. + 5.12 V_{\text{m/s}}} \] |
| RAYMODE (1976?) | \[ A = 2.10^{-6} f_{\text{Hz}} V_{\text{km}}^2 \] | \[ B = \max\{b_1, b_2\} \]
                             \[ b_1 = \sin \theta_{\text{rad}} - \left(\sin \theta_{\text{rad}} / \theta_{\text{rad}}\right) \left(\exp\left(-\frac{1}{4} a \theta_{\text{rad}}^2\right) / \sqrt{\pi a}\right) \]
                             \[ b_2 = \frac{1}{2} \sin \theta_{\text{rad}} \text{ with } \frac{1}{2 a} = \tan^2 \beta_0 = 0.003 + 0.0026 V_{\text{km}} \] |
| Leibiger (198?) | \[ A = 1.16 \times 10^{-6} f_{\text{Hz}} V_{\text{km}}^2 \] | \[ B = \min\{0.99, \max\{b_1, b_2\}\} \] |
| Weinberg (198?) | SL1 = 0                                                               | Leibiger’s loss if less than 11dB; 11dB else                                |

Table 1: The different versions of the Beckmann and Spizzichino model for surface loss.

Involved parameters are the frequency \( f \), the grazing angle \( \theta \), the reference wind-speed \( V \), the average peak-trough height \( H \) (\( H = 0.625 \times H_{1/3} = 0.625 \times 4 \times \sigma \), where \( \sigma \) is the r.m.s. surface height) and the variance \( \tan^2 \beta_0 \) of sea surface slope.
Fig. 2: Comparison of the Low Frequency term SL1 of B-Sp model with the set of Marsh and Schulkin experimental data (assuming Podeswa’s relation $H_{ft} = 2.04 \times 10^{-2} V_{km}^2$).

Fig. 3: Comparison of the High Frequency term SL2 of B-Sp model with its theoretical expression as 1-integral of Beckmann’s formula for very rough sea surface scattering strength.

<table>
<thead>
<tr>
<th>Lavor’s expression for $\Sigma$</th>
<th>Our understanding of Beckmann’s work</th>
</tr>
</thead>
<tbody>
<tr>
<td>$a = 1$</td>
<td>$a = \frac{1}{2}$</td>
</tr>
<tr>
<td>$F = \frac{1 + \cos \theta_1 \cos \theta_2 - \sin \theta_1 \sin \theta_2}{\cos \theta_1 + \cos \theta_2}$</td>
<td>$F = \frac{1 + \cos \theta_1 \cos \theta_2 - \sin \theta_1 \sin \theta_2 \cos \theta_3}{\cos \theta_1 + \cos \theta_2}$</td>
</tr>
<tr>
<td>$\phi = \frac{\sin^2 \theta_1 + \sin^2 \theta_2 - 2 \sin \theta_1 \sin \theta_2 \cos \theta_3}{(\cos \theta_1 + \cos \theta_2)^2}$</td>
<td></td>
</tr>
</tbody>
</table>

Table 2: Interpretations of the very rough sea surface scattering strength $\Sigma$ according to Beckmann: Lavor’s version ([6]) and our interpretation (We use Beckmann’s notations [5] for angular variables).
4. **EMPIRICAL DECAY RATES IN CONFINED CHANNELS**

Literature contains some representations of the surface attenuation not as a loss per bounce, but in terms of an attenuation excess $\alpha$, to be added to visco-chemical absorption.

Surface channels: Saxton & Baker:  

\[
\alpha_{\text{dB/kyd}} = \frac{26.6 \ f_{\text{kHz}} \ 1.4 \ \text{SeaState}}{\left(1452. + 3.5 \ T_{f} \ Z \ \right)^{1/2}} 
\]

$Z =$ channel depth  

\[
3.5\text{kHz}<f<7.5\text{kHz}
\]

In another paper ([7]) of the UAC 2014 conference, we review and interpret these formulas, which can also be used within their limited domain of applicability.

5. **CONCLUSION**

Despite decades of work, the problem remains quite open when time comes for us to adopt a pertinent model for surface loss in sonar performance assessment. We have reviewed and given the limitations of the most classical existing formulas. Future effort should be devoted to both theoretical and experimental investigations of the question. A theoretical reflection should focus on the capability of a simple reflection coefficient for describing scattering from surface roughness: such a description works only for the specular part of the reflected; the role of the scattered contribution is more complex and should at least imply a dependence of an effective reflection coefficient on transmitted waveform, on array geometry and on processing technique. More physically, the role of stable layers of air bubbles can be directly investigated (see [7]) and should be completed with an analysis of other populations of larger bubbles (associated with whitecaps) and also of the impact of fishes with swimbladders. Large scale experiments should otherwise have to examine variability of propagation loss in the same area, with the same source and receivers, over large periods involving as various sea states as possible.

**REFERENCES**


ANALYSIS OF SONAR DETECTION PERFORMANCE IN SOUTH CHINA SEA FOR ASW USING ASORPS

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Abstract: Objective of this study is to analyse the sonar detection performance under various temporal and space in South China Sea. This underwater environment is so complex because the ranging from 100m to 4000m water depth region contains internal waves occurring daily and the frequent typhoons passing through annually that creates the greatest noisy environment to conceal submarine from detection. The antisubmarine warfare (ASW) tactics is heavily dependent on sonar range prediction. For having a better understanding of how sonar performs in this region, this study uses sonar range prediction model (Advanced Sonar Range Prediction System, ASORPS) to carry out a simulation analyses. In ASORPS, we first use the range-dependent acoustic model with the hydrographic output of the Taiwan Coastal Ocean Nowcast and Forecast System (TCONF5) to map the sound speed profile distribution, bathymetry and sediment database to compute reverberation level, transmission loss and ambient noise levels, then calculate detection coverage and probability. A study which acoustic data measured in ASIAEX 2001 is used to compare with the simulation. This paper also presents the simulation results of the sonar detection performance in South China Sea. Conclusions show that sonar performance is good in autumn, night time and shallow water.

Keywords: sonar detection performance, detection probability, sonar range prediction
1. INTRODUCTION

The South China Sea is one of the most important shipping lanes on earth with over 1.6 million m³ (10 million barrels) of crude oil per day through the Strait of Malacca. Other than that, in terms of reserved resource, according to 2013 report by US Energy Information Administration, it points out the estimated oil reserve reaching to 11 billion barrels, natural gas reaching to 190 trillion cubic feet and the volume of the gas hydrate being around $3.2 \times 10^{12}$ m³ to $170 \times 10^{12}$ m³ in South China Sea [1].

The region this study focused is the north part of South China Sea where the depth is from 100m to 4000m. Since this area is next to the fringe of continental shelf, the topography is changeable and complex. The weather may change frequently. In winter, the north-east monsoon from Siberia plateau makes sea state worse. In summer, both of East Asian monsoon and South Asian monsoon would meet and interact at the same time [2]. There were 1,383 typhoons in North-West Pacific Ocean and South China Sea from 1958 to 2010, which was about the average of 26.4 per year. When typhoons formed in West Pacific Ocean, part of them route through Luzon Island, the South China Sea and finally land on Mainland China. The previous research has shown that the internal wave is formed due to the terrain factor that is quite close to Luzon Strait, which is between Batan and Sabtang Island [3]. When the internal wave transits toward continent, it is influenced by terrain resulting the difference on temperature and current in shallow water [4][5][6].

As for an island-like country, Taiwan heavily relies on energy import which is 99.25% of total requirement. The South China Sea locates in the south of Taiwan and plays an important role as shipping lane for most of those imports. The weather, hydrology, acoustic environment, and geology in this region are so complex that makes ASW difficult.

Underwater target detection is the key issue in ASW and the tactics is largely dependent on sonar range prediction. There are factors relate to sonar range prediction such as weather, ocean environment and parameters of sonar system. It is hard to predict accurate performance of a sonar system by a simple method. A computer-based analysis system equipped on ASW platform and command center aided to execute near real-time detection system is necessary. Then, this can provide the ability for fleet commands to handle all sonar system efficiency and performance in combat zone.

In order to get more accurate sonar range prediction for ASW operation, the Underwater Acoustics Laboratory (UAL) of National Taiwan University has been cooperated with ROC navy since 1996. Compare with simulation of the model and the measurement data around Taiwan year by year. After years of modifying and upgrading, the Advanced Sonar Range Prediction System (ASORPS) Version 3.0 is completed.

2. ASORPS

2.1. Sound Propagation Model

To calculate the sound wave energy below the water, the wave equation or Helmholtz equation used cylinder coordinate axis system $(r,\theta,z)$ are adopt. We also suppose that the ocean environment is axial symmetry, which the sound wave is not changed by $\theta$, then the
equation can be simplified to 2-dimentional. The FFP (Fast Field Program), Normal Mode and PE Approximation are developed from the equations. The ASORPS adopts two sound propagation models. In case the frequency is less than 1 KHz, the famous RAM module (Range-dependent Acoustic Model, RAM) from M.D. Collins Naval Research Laboratory is used [7]. The RAM module features that it can calculate a spread angle and can allow bigger trapezoid change in vertical direction to adapt to complex hydrology environment change. Otherwise the frequency is higher than 1 KHz, Gaussian Beam Model (GBM) is used [8]. The GBM is based on the ray theory. But it is different. The GBM includes limited sound ray of Guess energy distribution to calculate acoustic field and then superposition the energy. This can avoid energy diffuse from original ray theory.

2.2. Ocean Model & Sound Speed Profile

TCONFS is an experimental, real-time Ocean Nowcast/Forecast System for the waters off the coast of Taiwan. With an area of coverage extending from 18°N to 27°N and 117°E to 125°E, the system produces such daily nowcast variables as sea level variation, 3D ocean current, temperature, and salinity [9]. TCONFS is a 1/24 degree and 31 sigma level data-assimilating ocean model based on topography from the TaiDBMv6 bathymetry database of the Taiwan National Science Council Ocean Data Base (NSC/ODB). The model is restarted from previous nowcast measurements at -24 hours, and continuously assimilates daily microwave + infrared optimally interpolated sea surface temperature using Remote Sensing Systems as well as monthly climate temperature and salinity from NSC/ODB. Three hourly surface heat fluxes, solar radiation, wind stresses, and sea level air pressure from CWB/WRF are used for surface forcing [10]. Open boundary conditions (BC), including sea surface elevation, transport, temperature, salinity and current, are based on Hybrid Coordinate Ocean Model (HYCOM) [11]. The HYCOM consortium is a multi-institutional effort sponsored by the National Ocean Partnership Program (NOPP), as part of the U. S. Global Ocean Data Assimilation Experiment (GODAE). Barotropic tidal forcing is applied to the open boundary grid by superimposing tidal elevation and transport for 11 tidal constituents (Mm, Mf, K1, O1, P1, Q1, K2, M2, N2, S2, and M4) on the (non-tidal) BC from HYCOM using a forced radiation BC. Tidal data is taken from the Oregon State University global tidal database [12].

2.3. Ambient Noise Model

Sea surface shipping and wind wave are main sources of underwater background noise [13]. The ASORPS uses the Acoustic Module for Sea-surface Noise (AMSN) developed by National Taiwan University to research ambient noise [14]. There are four modules in AMSN, including Source Strengths, Noises Distribution, Acoustic Fields, and Ocean Environment prospectively. The first two parts relate to noise types, such as noises from ships have relations on shipping density, whereas waves have relations on surface wind speed. Therefore, they can be used as initial conditions. We assume the model is limited on horizontal range. And a total of N underwater acoustic sources distribute in the sea as background noise. The latter two adopt ocean module to provide sound speed as the underwater noise propagation conditions. The sound propagation uses RAM 1.5 version, and the noise frequency is set to 125 Hz for wind wave.
2.4. Reverberation

Reverberation Level calculation using Lambert’s Law [15], considers parameters of sediment, sonic frequency, beamwidth and other parameters in empirical formula to estimate sea bed reverberate. The results can be found that in sand bottom area appears strong reverberation signals, whereas in the mud bottom regions are observed weak reverberation signals.

2.5. Bathymetry & Sediment

Bathymetry was extracted from Marine Environmental Databank (MED) by Taiwan Ocean Research Institute (TORI), and the geoacoustic sediment description was calculated using the Hamilton-Bachman method with data from Naval METOC Office R. O. C. [16].

3. ASIAEX

3.1. The Experiment

Asian Seas International Acoustics Experiment (ASIAEX) was conducted from April to August in 2001, which was multinational collaborated scientific project including the experiment in South China Sea. The geophysical location on continental shelf (117°E~117.5°E, 21.5°N~22.2°N) of the South China Sea Component of ASIAEX is shown in Figure 1 (a).

![Fig. 1: (a) Locations of ASIAEX 2001 and ASORPS simulations. (b) Detection range affected by internal tide and nonlinear internal wave in ASIAEX 2001 and compare with ASORPS simulation in same location.](image)

The instruments were deployed across and along the shelf. The VLA/HLA listened acoustic signal from deep southern sources and shallow eastern source, which corresponding to across-shelf path and along-shelf path respectively. The southern sources were deployed in deeper water (~350m) and transmitted with 224Hz and 400Hz signals.
These sources transmitted acoustics signals during the time from 30 April to 20 May, covering one and half cycle of the spring tide and the neap tide [17].

In addition to the acoustic instruments, there are several environmental moorings deployed in the test area to record temperature, salinity, pressure and the current.

3.2. Comparison With ASORPS

Based on study of Yuan et al., the ASIAEX 2001 experimental record of May 4 and May 11 measured temperature data has been used to calculate the sonar detection range, and to analyze the impact of detection range during internal tide (IT) and nonlinear internal waves (NIW).

Five days average temperature taken in case 1, and 3 hours average temperature taken in case 2 for fitting the sound speed and calculated the detection range. The propagation path was the same track as the arranged experimental instruments. The results are shown in Figure 1 (b). It can be seen the large variation of detection range in the NIW impact on May 11 [18].

To compare the sonar detection range of ASORPS with Yuan’s, the ASORPS also does the analyses at the same area 3 hours average SSP per day of May by TCONF. We could see the periscope depth range (PDR) and best depth range (BDR) change slightly over time. It is also similar with 2-IT of IT impact on May 4, and the shorter detection range due to different parameter setting.

4. SIMULATION

The ASORPS analyzes the signal excess (SE) of active and passive sonar according to the sonar equation. Also, it shows the cumulative normal distribution of probability with relations between SE and detection probability based on the basic detection theory [13]. When SE is 0 dB, the detection probability is 50%. Whereas when SE is 30dB, the detection probability is 100%. So as to the SE is -30dB, the detection probability becomes 0%.

We assume sonar and target parameters in Table 1 in the area of 18°-22°N, 117°-119°E northern waters of the South China Sea as show in Figure 1 (a). The prediction is calculated with grid points 1/12 ° (approximately 5 nautical miles). The total grid point is 1,200. And each point only calculates one radical direction which is 000°. The target is set at the distance of 5000m, depth of 18m. The detection probability can be solved from SE, to analysis the sonar detection probability in South China Sea.

<table>
<thead>
<tr>
<th>SL (dB)</th>
<th>Freq. (Hz)</th>
<th>Source Depth (m)</th>
<th>Beam Width (degree)</th>
<th>Target Depth (m)</th>
<th>DI (dB)</th>
<th>TS (dB)</th>
<th>DT (dB)</th>
</tr>
</thead>
<tbody>
<tr>
<td>220</td>
<td>5000</td>
<td>8.5</td>
<td>30</td>
<td>18</td>
<td>10</td>
<td>10</td>
<td>10</td>
</tr>
</tbody>
</table>

*Table 1: The main parameters of ASORPS inputs*
The detection probability in each season is as Figure 2 showing the active sonar performance in four seasons. In spring (April) and summer (July), deep blue occupies the most area implying the detection probability is less than 30%. As in autumn (October) and winter (January), the detection is much higher than the other two seasons in average. The best detection probability appears in autumn because of the warmer than air water being around. Since the surface layer tends to be cooler than the lower layer, it could expect an equal or weaker positive gradient above the main thermocline that is as the sound speed profile of Figure 2 (a) and (d). There are more rays will be deflected upwards and could create sound channel near the surface, in where it can keep more sound energy, that is the reason for having higher detection probability in cold weather.

To compare with autumn and winter, according to the measurements of average wind speed on Tung-Sha Island in 1988-2004, it’s 8 kts in autumn and 15 kts in winter. The wind in winter is much stronger than autumn. When convection in water brings higher ambient noise and decreases SE in winter, the sonar performance in autumn is then much better.

On the other hand, summer’s higher surface temperature and propagation loss would cause lower EL. In the end, it is hard to detect underwater targets due to the negative gradient and the shielding of reverberation.

In addition, there is an interesting observation in the results from Figure 2, which is at isobaths of 800m. The detection probability is lower than 20% in most deep water areas. However in the depth of 800m or lower in northwestern edge of the continental shelf, the detection probability is high than 80%. It is obvious that the dramatic changes here are relevant to topography.

Meanwhile, we set the sonar type to be cylindrical spreading in simulation; the acoustic energy propagation horizontally keeps more energy in shallow water. The sound rays also reflected smoothly on the sandy bottom. It results higher detection probability in continental shelf area.

Fig. 2: Active sonar detection probability and sound speed profile of four seasons at South China Sea simulated by ASORPS

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Specifically, the detection probability averages up to 80% in each season at the corner of northwestern continental shelf break waters nearby 22°N. It might be associated with the local sandy sediment.

For purpose of observing the variation of detection range in a single day, ASORPS has 4 hours average SSP of May 12, and then has it simulated. The results shown, in Figure 3, the minimum detection range is at time 09-12 hours and the maximum detection range occurs at time 01-04. In general, the night time has longer range of detection than in the day time. It is due to the effects of sea surface temperature, which is same as the analyses to seasons.

![Fig. 3: The periscope depth range and best depth range of South China Sea on May 12.](image)

Another observation of the distribution of long-term SSP in the South China Sea is that it can be found in most of the SSP changes only within the depth 500m, and almost the same in deeper water for temporal and space variations. The lowest sound velocity point is at about 1000m where the negative gradient goes to positive and the sound channel existed.

5. CONCLUSIONS

ASORPS can calculates the sonar performance in different acoustic field, and transit it to the detection probability by statistic methods. It becomes valuable when used in the ASW decision making process which may provide commander the useful information in searching the threatened submarine in this complex underwater battle field.

The findings in this study are as follows. 1. The sonar detection probability in the autumn is greater than it is in the spring. 2. The sonar detection performance in shallow water is greater than in the deep water. 3. The night time has longer range of sonar detection than in the day time.

Most of submarines operate within depth of about 500m, with which this study also shows that the sonar performance is affected in temporal and space due to rapid ocean environment. From perspective of ASW, when a submarine transits close to the surface while away from the continental shelf area, during the day time can be safe from the detection of rivalry

6. ACKNOWLEDGEMENTS

The authors deeply appreciate Professor Kuei-Min Wang and Professor Mei-Chun Yuan for their valuable recommendations.
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SEDIMENT ACOUSTICS: THE NEED FOR IMPROVEMENT

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Abstract: The high frequency environmental acoustics sediment model published in the High-Frequency Ocean Environmental Acoustic Models Handbook (APL-UW 9407), which has been widely adopted by underwater acousticians and sonar modelers, is examined in the light of recent sediment acoustic models and measurements. Due to multiple scattering effects the sound speeds and attenuations for the larger grain sizes ($\phi$<1) need to be updated. The sediment densities for the middle range of grain sizes (1<$\phi$<5) are underestimated due to poroelastic effects. Sound speed dispersion and frequency dependence of attenuation are over simplified. A poroelastic model with extensions to account for grain contact physics is proposed. For practical applications, an efficient parameterization of the poroelastic model allows the number of adjustable parameters to be reduced to a level comparable with that of simpler fluid and elastic models, while retaining all its physical advantages.

Keywords: sediment acoustics, bottom loss, poroelastic, grain shearing.
1. INTRODUCTION

Active sonar performance in shallow water is significantly dependent on the acoustic properties of the seabed. This is true of a wide range of sonar systems, and over a wide range of frequencies. Therefore it is important that the acoustic properties of the seabed be properly understood and modeled. The bottom forward loss has an impact on the propagation loss of the signal to and from targets of interest, and the bottom scattering strength directly impacts the reverberation that interferes with the desired signal. The forward bottom loss also impacts the performance of passive sonars since it determines the received signal level and its relationship to the ambient noise. Any shallow water sonar performance prediction model must have a model of the acoustic properties of the seabed. In this study, a number of existing models are considered. The problem areas are described and possible improvements are put forward.

2. FLUID, VISCOELASTIC SOLID MODELS

The High Frequency Environmental Acoustic (HFEVA) model version 1.0, which is described in the APL/UW 9407 report [1], and its successor the Geoacoustic Bottom Interaction Model (GABIM) [2], are fluid and viscoelastic solid models. Both models were developed by researchers at the Applied Physics Laboratory, University of Washington (APL/UW). For mud sediment types, they are adequate but for sandy sediments they have a number of problems. They are based on the assumption that the density ratio \( \rho_s / \rho_w \), sound speed ratio \( c_p / c_w \), and log decrement \( \delta_p \) in the sediment are functions of mean grain diameter \( d \), and independent of frequency. From the definition of the log decrement, it is implicit that the attenuation \( \alpha_p \) increases linearly with frequency \( f \). These assumptions are based on trends that were observed in measured sound speed and attenuation values, such as those reported by Hamilton [3] up until about the 1980s.

\[
\rho_s = \rho_w \alpha_p(d) \tag{1}
\]

\[
c_p = c_w \nu_p(d) \tag{2}
\]

\[
\alpha_p = \frac{\delta_p}{c_p}(d) \tag{3}
\]

In the above equations, the subscript “w” refers to water, and “p” to the compressional wave in the sediment. Given that the inverse-Q is defined as,

\[
Q_p^{-1} = \frac{\alpha_p c_p}{\pi f} \tag{4}
\]

Substituting from Eqs. (3) into (4), it is evident that the inverse-Q must be independent of frequency.

\[
Q_p^{-1} = \frac{\delta_p(d)}{\pi} \tag{5}
\]

Similar expressions for the shear wave are obtained by replacing the “p” subscript with the “s”. The models are simple, containing 3 parameters in the fluid case. In the viscoelastic case, there are two more: shear speed ratio and shear log decrement.

There is a theoretical problem with the above formulation. It is inconsistent with the Kramers-Krönig relationship, which is applicable to causal processes. The Kramers-
König relationship may be expressed in terms of an integral relationship between the wave speed and attenuation [4].

\[ c_p(\omega) = \frac{c_p(\omega_0)}{\pi} \int_{\omega_0}^{\omega_1} \frac{Q_p^{-1}}{\omega} d\omega, \quad \omega = 2\pi f \] (6)

This integral shows that for non-zero attenuation, i.e. non-zero inverse-Q, the wave speed \( c_p \) must change with frequency. This problem is overcome in the grain shearing (GS) model, which was developed by Buckingham in 2000 [5]. The compressional and shear wave speeds and attenuations are given by,

\[ \frac{1}{c_p} = \frac{1}{c_o} \Re \left[ 1 + \frac{3\gamma_p + 4\gamma_s}{3\rho_o c_o^2} (j\omega T)^n \right]^{1/2} \] (7)

\[ \alpha_p = -\frac{\omega}{c_o} \Im \left[ 1 + \frac{3\gamma_p + 4\gamma_s}{3\rho_o c_o^2} (j\omega T)^n \right]^{1/2} \] (8)

\[ \frac{1}{c_s} = \sqrt{\frac{\rho_o}{\gamma_s}} \Re \left[ (j\omega T)^n \right]^{-1/2} \] (9)

\[ \alpha_s = -\omega \sqrt{\frac{\rho_o}{\gamma_s}} \Im \left[ (j\omega T)^n \right]^{-1/2} \] (10)

The GS model can be shown to satisfy the Kramers-König relationship and it is also a constant-Q model, therefore, it is equivalent in functionality to the HFEVA and GABIM models but without the theoretical drawback of non-causality. It also has the same number of parameters, i.e. 5, but they are different: \( c_o, \rho_o, \gamma_p, \gamma_s \) and \( n \). The parameter \( T \) is always set to 1 s, and does not count as an input parameter.

3. THE NEED FOR IMPROVEMENT

There is a more practical problem with the models, and that is the measured and inverted sound speed in sandy sediments, that have been collected since about the 1980s, show a bigger change with frequency than the models predict, and the attenuation does not always increase linearly with frequency, as shown in Figs. 1 and 2.

![Fig.1: Measured and inverted sediment acoustic attenuation as a function of frequency from a number of references compared to the HFEVA and GS models.](image)
The measurements come from a wide range of experiments, including published low-frequency measurements that were collected into a single publication by Zhou, Zhang and Knobles [6], high frequency measurements by Turgut and Yamamoto [7], and by the participants in the Sediment Acoustics Experiments of 1999 [8] and 2004 [9]. The post 1980 data are superimposed on the pre-1980 data from Hamilton shown as grey circles for comparison. The HFEVA model is often presented as a set of 23 default types, two of which (number 9 and 18) are shown in the figures as illustrations. The GS model shown is based on parameter values from Ref. 5. It is clear that the trends in the data and models do not agree, particularly at frequencies below about 3 kHz.

In response, the GS model was modified to better fit the measurements. The modified model was called the viscous grain shearing (VGS) model [10]. This model has one more parameter than the GS model. The additional parameter $\tau$ represents the time constant of a relaxation process that causes the attenuation to deviate from the constant-Q trend at low frequencies and provide better agreement with the measurements, shown in Figs. 3 and 4.
Even though the VGS model was able to somewhat match the sound speed and attenuation in the sediment, it did not match the measured shear wave speed at 1 kHz, as pointed out in Ref. [11]. In response, a further modification of GS, called VGS($\lambda$) was put forward [12], which had another relaxation process and its associated time constant, which brings the total number of parameters to 7. However, both VGS and VGS($\lambda$) predict a rather unphysical behavior regarding the shear wave. They both predict that the shear wave speed goes to zero in the low-frequency limit, at a rather rapid rate that is inconsistent with the known behaviour of shear waves in sand.

4. POROELASTIC MODELS

A fundamental issue with HFEVA, GABIM, GS, VGS and VGS($\lambda$) is that they are all either fluid or viscoelastic models. No matter how the predicted wave speeds and attenuations change with frequency – even if they fit the measured values - they still only have one compressional wave and one shear wave. Consequently, the reflection coefficient at the water/sediment interface is still governed by the boundary conditions between a fluid and an elastic medium. It has been shown by Williams [13] that such models predict reflection losses that are inconsistent with measurements, as shown in Fig. 5. To fit the reflection loss measurements, it is necessary to use a poroelastic model, such as the Biot-Stoll model [14]. The poroelastic models have a second compressional wave, which alters the boundary conditions at the fluid/sediment interface. Although the second compressional wave, also known as the Biot slow wave, is highly attenuated in water-saturated sands, it is a significant loss mechanism that materially changes the reflection loss. Included in this category is the effective density fluid model (EDFM) by Williams [8]. This is a reduced Biot-Stoll model in which the frame moduli are set to zero. This model is a fluid model, because it does not support a shear wave, but it supports two compressional waves and is able to match the measured reflection loss almost exactly like the Biot-Stoll model in surficial water-saturated sand without any overburden pressure. To obtain a poroelastic model with the proper frequency dependence, it is necessary to include the grain-grain contact physics, such as the models in Refs. 15 and 16, and later...
developments thereof. The main drawback of the Biot model, and variations thereof, is that it has a large number of input parameters. Recognizing that some of the parameters only change within a very narrow range of values, and using alternative parameterizations, it is possible to reduce the number of active input parameters to 7, which is comparable to the number of parameters in the VGS(λ) model.

![Fig.5: Measured reflection loss over a sand sediment as a function of grazing angle compared to fluid, viscoelastic and the Biot-Stoll models, adapted from Fig. 10 of Ref. 13.](image)

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REFERENCES

DYNAMIC SELF-ORGANIZING ALGORITHM FOR UNSUPERVISED SEGMENTATION OF SIDESCAN SONAR IMAGES

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Abstract: This paper deals with the dynamic neuronal approach for segmentation of textured seafloors from sidescan sonar imagery. For classical approaches of sonar images segmentation, the result of the classification is a set of sediment clusters representing the different kinds of seabed. However, those classical approaches give satisfying results only when a comprehensive training set is available. If the training set lacks a particular kind of seabed, it will be unknown for the classifier and the classification will be reduced to the closest known sediment cluster. As it is not always feasible to know the entire seabed types before the training phase, a dynamic algorithm solution capable of incremental learning has been developed. The Dynamic Self-organizing maps (DSOM) algorithm used in this work is an extension version of classical SOFM (Self-Organizing Feature Map) algorithm developed by Kohonen combined with Adaptive resonance Theory (ART). It is based on growing neuronal map size during the learning processes. Therefore, the size of the map is small in the beginning but increase dynamically using control vigilance threshold. To assess the consistency of the proposed approach, the DSOFM algorithm is tested on simulated data clusters and on real sonar data.

Keywords: Unsupervised clustering, SOFM (Self-organizing feature maps) algorithm, ART (Adaptive resonance Theory, DSOFM (dynamic self-Organizing feature maps), texture analysis, sidescan sonar images.
1. INTRODUCTION

Image segmentation is an important step in image analysis chain with different applications to image processing, pattern recognition and objects detection, etc. Segmentation algorithm consists on the process of image division into homogenous groups of pixels according to statistical measure similarity. Two families of segmentation algorithms can be distinguished: the supervised and the unsupervised approaches. In the supervised algorithms a priori knowledge is needed to get a successfully results. The most important supervised algorithms use Maximum Posteriori (MAP) or Maximum Likelihood (ML) techniques [1].

Like with other types of images, supervised algorithms of sonar images segmentation use ground truth, acquired at discrete locations by video, dredge or core data sampling, to assign labels to the seabed types. The supervised approach gives satisfying results only when a comprehensive training set is available. If the training set lacks a particular kind of seabed, it will be unknown for the classifier and the classification will be reduced to the closest known sediment class. As it is not always feasible to have seabed ground truth classes and to know the entire seabed types before the training phase, an unsupervised algorithm capable to detect clusters according to statistical similarity and independently to the expert interpretation is suitable for sonar images. This is what, automated sonar systems classification are becoming widely used. Recent progress in underwater robotics has led to the development of autonomous underwater vehicles (AUVs) capable of automatic data collection and interpretation. The onboard processing capability of these AUV allow for real-time implementation of algorithms for unsupervised seabed classification.

The unsupervised approaches exploit the resemblance between statistical features estimated from images, with any a priori knowledge about data labeling or number of group. In this case, clustering algorithms are used to gather pixels or regions on similar groups.

Approaches to unsupervised learning include: clustering algorithms (e.g., isodata, k-means, mixture models and hierarchical clustering) [2][3], blind signal separation generally used for dimensionality reduction and features extraction (e.g., Principle component analysis (PCA), Independent component analysis (ICA)) [3] and neural network models that using unsupervised learning. Among these models, Self-Organizing Feature Maps (SOFM) developed by Kohonen [4] and Adaptive Resonance Theory (ART) developed by Carpenter and Grossberg [5]. Several works have applied successfully different approaches of artificial neural network (ANN) to the problem of seafloor classification [6][7][8].

In this work, a new approach for unsupervised segmentation of sidescan sonar images is proposed. Our approach is based on the mixture of two neural network algorithms: the SOFM and ART algorithms.

The SOFM algorithm is powerful tool for clustering and Data Mining. It has been used for mapping high-dimensional data into generally one, two or three dimensional feature map [4]. The most important characteristic of SOFM algorithm consists on topology preservation of input space using neighbored function. It means that data of input space which are close in term of features distance will be close after projection by SOFM algorithm. This topology preservation of data allows best visualization and identification of data clusters. The SOFM algorithm is normally presented as two-dimensional (2D) grid
of neural nodes. A group of close nodes of the grid represent a given class cluster of the data. However, classical SOFM algorithm has some limitations. One of these problems is that the size of the grid and the number of nodes have to be predetermined. Therefore, more simulations tests must be conducted to define the appropriate size of the map. In the case of unknown structure of the data, an incremental or dynamic structure of the grid is suitable.

Several dynamic neural network models have been developed, which attempt to overcome the limitation of the fixed size grid of the classical SOFM algorithm. The Neural Gas Algorithm (NGA) developed by Martinez and Shulten [9] is an unsupervised neural network. The main idea of this algorithm is to successively add a units (or nodes) to an initial small network by evaluating local statistical measures gathered during previous adaptation steps. Another Algorithm called Growing Cell Structures (GCS) developed by Fritzke [10] based on the same approach of NGA. However, the GCS has a fixed topology dimensionality (2-D or 3-D). Alhakoon et al [11] proposed a dynamic Self-organizing Maps with controlled growth (GSOM) for knowledge discovery. The advantage of GSOM is the control of the size of the grid using spread factor. The spread factor in this case is independent of data dimensionality and can be used as threshold to create different maps with different dimensionality.

A dynamic Self-organizing feature map algorithm (DSOFM) is proposed in this paper. The unsupervised algorithm proposed is based on the combination of SOFM and the ART Algorithms. A detailed description of this algorithm is presented in the section 2. Section 3 presents some experiments and discussion of the DSOFM algorithm tested on synthetic data and on real sonar data. The section 4 gives a conclusions and recommendation of this work.

2. DYNAMIC SELF-ORGANIZING MAPS (DSOM)

The proposed algorithm in this work is based on the combination of two neural networks: SOFM and ART algorithms. The principle idea is to initialize a grid map with small number of nodes then a threshold based on the vigilance parameter of ART algorithm controls the growing size. Each input presented to the network is compared to each node of the network. If resonance is occurred, it means the presented input is matched to one of the nodes, then no growing of the grid. However, if the input is so far in term of Euclidean distance to the all nodes, new nodes are created and the size of the grid is extended. During the node growth, the weights values of the nodes are updated according to the learning process of classical SOFM.

The proposed DSOFM is base on three major phases: initialization, growing phase and stopping step. The process is as follows:

A. Input data processing

The input preprocessing option concerns normalization. It is shown below that input normalization prevents a problem of category proliferation (more clusters) that could otherwise occur [12]. A normalization procedure called complement coding [5] is used.

Each input vector \( x \) is a p-dimensional vector \( x(x_1, x_2, \ldots, x_p) \).
0. **Normalization**: Each component \( x_i \) is in the interval \([0,1]\)

1. **Compute the complement code** \( x^c \) of \( x_i \):
   \[
   x^c_i = 1 - x_i 
   \]  

   The new input is 2p-dimensional vector given by:
   \[
   X = (x, x^c) = (x_1, x_2, \ldots, x_p, x^c_1, x^c_2, \ldots, x^c_p)
   \]

B. **Initialization phase**

The network is initialized with nine nodes (a grid of 3x3) with random values from the input vector space. The choice of this number of nodes to initialize the network is justified to implement a 2-D lattice structure and each node has at least two neighbors.

C. **Growing phase**

0. A vector \( x_i \) is chosen randomly presented to the input of the network.

1. **Calculate the activated neuron of the presented input to neurons using**:
   \[
   T(j) = \frac{|X(i) \land w(i)|}{|w(i)| + \beta} 
   \]

   \( \land \) is the fuzzy AND operator \([13]\) defined by:
   \((x \land y) = \min(x, y)\); and where the norm \( |.| \) of a given vector is defined by:
   \[
   |X| = \sum_{i=1}^{2M} X_i 
   \]

   \( X(i) \): is an input from vector space.
   \( w(i) \): is weight vector.
   \( T(j) \): presents the activated neurons.
   \( \beta \): the bias defined in ART algorithm \([5]\), this value must be within the range \([0, 1]\) (although values very near to zero are best).

2. **Selection of the winner neuron** \( j^* \) **from the activated neurons**:
   \[
   w(j^*) = \text{Max}(T(j))
   \]

3. **Growing process with vigilance threshold**:
   \[
   \text{If } \frac{|X(i) \land w(j^*)|}{|w(i)|} \geq \rho 
   \]
then
\[ \text{node} = \text{node} + 3 \]

\[ w(i)_{\text{new}} = w(i)_{\text{old}} + \alpha \cdot V(i) \cdot [X(i) \wedge w(i)] \] (4)

\[ w(i)_{\text{new\_node1}} = \alpha \cdot X(i) + (1 - \alpha)w(i)_{\text{winner1}} \]
\[ w(i)_{\text{new\_node2}} = \alpha \cdot X(i) + (1 - \alpha)w(i)_{\text{winner2}} \]
\[ w(i)_{\text{new\_node3}} = \alpha \cdot X(i) + (1 - \alpha)w(i)_{\text{winner3}} \]

\[ w(i)_{\text{new}} = w(i)_{\text{old}} + \alpha \cdot V(i) \cdot [X(i) \wedge w(i)] \] (4)

\( \rho \): is the vigilance parameter, this value must be within the range \([0, 1]\).
Low vigilance value minimizes a number of clusters and inversely a high value allows clusters proliferation.
\( \alpha \): is the learning rate (\( \alpha \in [0, 1]\)).
\( V(i) \): neighbourhood function of SOFM algorithm (Gaussian function is very often used).
\( \text{node} \): is the number of nodes of the network, initialized by a grid of 3x3 size.

4. Return to step 0 as all samples are presented to the network.

3. DISCUSSIONS OF EXPERIMENTS

The subject of clustering algorithms is to discover a grouping of structures inherent in data. In this section experimental tests are used to show the capability of DSOFM algorithm for incremental clusters discovery. Two types of experiments are conducted. The first one is the application of the DSOFM algorithm for clustering on simulated synthetic data. The second experiment is devoted to test DSOMF algorithm on real sonar dataset.

A. Experiment 1

The first data used for experiment is simulated of two datasets shown in Figure.1. The dataset in the right of the Figure.1 contains 7 clusters, 788 vectors, in 2-dimensions and the second contains 399 vectors, in 2-dimensions with 6 clusters. In the Figure.2 four random Gaussian data generated in interval \([0, 1]\) are used to test DSOFM algorithm.
Fig. 1: Clustering of two simulated data using DSOFM algorithm.

Fig. 2: Example of application of DSOFM algorithm for simulated Gaussian data with different means and standard deviations. A) 2 Gaussians, B) 3 Gaussians and C) 4 Gaussians.

In the Figure 1, the result of the application of DSOFM algorithm to the two datasets demonstrates the ability for cluster discovering. The same observations is shown in the Figure 2, in all cases (2, 3 or 4 Gaussians) we show the deployment of the DSOFM neural network to discover the different clusters.

A. Experiment 2

The sonar data used for our study were obtained during the BP’02 (Battlespace Preparation) experiments carried out by the SACLANT Undersea Research Centre in La Spezia, Italy. The system used is the Klein 5000 sidescan sonar operating at 455 kHz. For experiment and to assess the consistency of DSOFM algorithm, a data base of 400 images of four types of sediment (Posidonia oceanica, rock, ripples and Sand) is created from the sonar data images used.

In the Figure 3, for a good comprehensive representation in 3-dimensional space of sonar datasets, only three features from Haralick [14] attributes are used: correlation, maximum of distribution and elongation factor.
In the Figure.3, four type of seabed (Sand, Rock, Vertical Ripples and Posidonia) are introduced the DSOFM network gradually. Dynamically the new seabed type added is detected by the DSOFM network by increasing his grid size (number of nodes) and adapted his structure. In the first case of only sandy and rocky seabeds are presented to the DSFOM network, the size of the grid is 4X3 neurones. Then the size of the network becomes 5X3 neurones for the given seabed (sand, rock and vertical ripples). Finally, the size of the network for the all 4 seabeds is 7x3 neurones.

4. CONCLUSIONS AND ONGOING WORK

This paper has presented a new dynamic approach for sidescan sonar images segmentation and classification. It is based on the combination of self-Organizing feature Maps and Fuzzy ART algorithms. The proposed approach gives good results in the specific case of a given data set. However, the objective is to assess the robustness of the algorithm by processing more comprehensive datasets. This will be achieved by processing existing datasets and by
implement a real time version of the classification algorithm for the future autonomous missions of the Daurade AUV.

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REFERENCES

PERFORMANCE ANALYSIS OF SINGLE RECEIVER MATCHED-MODE PROCESSING FOR SOURCE LOCALIZATION

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Abstract: Matched-Mode processing (MMP) is an alternative to Matched-Field processing (MFP) that explicitly uses the modal description of the acoustic field to estimate source position and/or oceanic environmental parameters. MMP involves two steps, first the modes are separated, and then they are compared to mode replicas in order to infer the desired parameter values. If the source emits a short time broadband signal, modal separation can be achieved with a single hydrophone using a time-frequency analysis tool called warping. Within this framework, a method that allows to predict MMP performance is provided: the Ziv-Zakai bound is derived, which gives a bound on the achievable performance that is much tighter than the Cramer-Rao bound. The results are then used to perform quantitative performance analyses of the single receiver MMP source localization problem. Particularly, performance with respect to noise is examined for incoherent MMP.

Keywords: Matched-Mode processing, maximum-likelihood, performance analysis, Ziv-Zakai bound
1. INTRODUCTION

Matched-Field Processing (MFP) allows estimation of source position and/or environmental parameters by maximizing a cost function called *processor* that quantifies the match between the recorded acoustic field and simulated replicas of the acoustic field. At low-frequency in shallow water, the propagation can be described by normal mode theory, and the acoustic field consists in several propagating modes. If the modes can be properly extracted from the received signal, MFP can be replaced by Matched-Mode Processing (MMP) [1,2]. Unlike MFP that considers the total pressure field, MMP explicitly compares the measured modes to simulated mode replicas in order to infer the desired parameter values. Obviously, extracting the modes from the received signal is a critical step in MMP. It usually requires the use of a dense vertical array spanning the whole water column [1,2]. However, the use of warping processing allows to filter the modes with a single receiver when the source is impulsive [3,4], which subsequently opens the door for single-receiver MMP (SR-MMP) [5].

Prediction of SR-MMP achievable performance remains an open question. A brute-force way to approximate this performance is to resort to Monte-Carlo simulations but this comes at the price of very time-consuming computer simulations. An alternative approach is to derive lower bounds on the mean-square error (MSE). A wide range of bounds has received attention in the underwater acoustics literature, including the Cramer-Rao bound, the Barankin bound, the Weiss-Weinstein and the Ziv-Zakai bounds [6]. In this study, the performance of maximum likelihood (ML) SR-MMP is analyzed through the Ziv-Zakai bound (ZZB). This choice is motivated by several relevant properties satisfied by this bound. First, the ZZB is known to yield tight and reliable error prediction at all SNR regions. In addition, the ZZB allows analyzing performances through the MSE or through the probability of outage error, which is the probability that the estimation error exceeds a given threshold. Knowledge of such a probability can be of interest for localization applications where one wants to know the probability of correctly locating a source within an acceptable error. The article is organized as follows. The single receiver matched-mode processing problem is introduced in Section 2. The ZZB is presented in Section 3. Finally, numerical examples applied to a source localization application are presented in Section 4.

2. SINGLE RECEIVER MATCHED MODE PROCESSING

2.1. Modal propagation and modal filtering

Low-frequency sound propagation in shallow water can be described by normal-mode theory. Considering a broadband source emitting at depth $z_s$ in a range-independent waveguide, the pressure field $y(f)$ received at depth $z_r$ after propagation over a range $r$ is given by

$$ y(f) = s(f) \sum_{m=1}^{M} x_m(f), $$

where $s(f)$ is the source spectrum and $M$ is the number of propagating modes. The quantity $x_m(f)$ is the contribution of mode $m$ to the pressure field. It is given by
where $k_m$ and $\psi_m$ are respectively the horizontal wavenumber and modal depth function of the mode $m$, and $Q$ is a constant factor. To perform MMP, it is necessary to filter the modes $x_m(f)$ from the received signal $y(f)$. When the source emits a short-time signal, this operation can be done with a single receiver using signal processing transformations known as warping operators. Detailed demonstrations of single receiver modal filtering can be found in [3,4,5]. In the next sections, it will be assumed that the modes are perfectly filtered. It is out of scope of this paper to properly study the impact of modal filtering on MMP performance. However, any degradation resulting from modal filtering can be embedded into the noise component contaminating the isolated modes.

### 2.2. Data model

The goal of SR-MMP is to estimate an unknown parameter set $\theta$ from the modes recorded on a single hydrophone. The parameter set $\theta$ may contain both source position (range, depth), and/or ocean environmental parameters. Consider $M$ modes received on a single receiver. The complex received signal is modeled by the $M \times 1$ vector

$$y(f_k) = s(f_k) \cdot x(f_k, \theta) + w(f_k), \quad k = 1, ..., K$$

where,

- $K$ is the number of available frequencies $f_k$.
- $x(f_k, \theta)$ is a complex $M \times 1$ vector representing the contribution of $M$ modes at frequency $f_k$. The $m$th component of $x(f_k, \theta)$ is given by Eq. (2).
- $s(f_k)$ is a deterministic complex scalar representing the source signal.
- $w(f_k)$ is a complex $M \times 1$ vector representing a circularly-symmetric, zero mean, white Gaussian noise with power $\sigma_w^2(f_k)$, and is independent of the source signal.

### 2.3. Incoherent matched-mode processor

Similarly to MFP, a common approach for deriving matched-mode processors is to resort the maximum likelihood (ML) framework. ML estimation presents the advantages of offering the best asymptotic properties with respect to efficiency and can be applied for any state of source spectral knowledge. In the case where no information is available on the sequence $s(f_k)$, the ML estimate of the desired parameter set $\theta$ is

$$\hat{\theta}^{\text{inc}} = \underset{\theta}{\arg \max} \quad C^{\text{inc}}(\theta), \quad \text{with} \quad C^{\text{inc}}(\theta) = \sum_{k=1}^{K} \left| \frac{\overline{x}(f_k, \theta) y(f_k)}{\sigma_w^2(f_k)} \right|^2,$$  

where $\overline{x}(f_k, \theta)$ is a normalized version of $x(f_k, \theta)$ defined as $\overline{x}(f_k, \theta) = \frac{x(f_k, \theta)}{\|x(f_k, \theta)\|}$. This processor is referred to as incoherent because the multiple frequencies are summed incoherently, i.e., with the absolute value that cancels phase information inside the double sum. In the following we will investigate the performance of this processor.
3. ZIV-ZAKAI BOUND

The ZZB is a Bayesian bound that provides a lower limit on the MSE averaged over the \textit{a priori} parameters probability density function. Not only it approximates the performance of Bayesian estimators, such as MAP processors, but it also approximates the MSE of deterministic estimators on the prior space when uniform prior is assumed. The basic idea behind the ZZB is to connect the estimation error to a simpler related problem in detection theory. The ZZB is obtained by lower-bounding the probability of outage, i.e., the probability that the estimation error fall above a given threshold. Define the estimation error as $\varepsilon = \hat{\theta}(y) - \theta$. In the scalar parameter case, the probability of outage $P_t\left(|\varepsilon| \geq \frac{h}{2}\right)$ is lower bounded as [7]

$$P_t\left(|\varepsilon| \geq \frac{h}{2}\right) \geq V\left\{ \int_{-\infty}^{\infty} \min(p_o(\theta), p_o(\theta + h)) \cdot P_{\text{min}}(\theta, \theta + h) d\theta \right\},$$

where $p_o(\theta)$ is the \textit{a priori} parameters probability density function, $V$ is the valley filling function defined by $V\{f(h)\} = \max_{\xi \in 0} f(h + \xi)$ and $P_{\text{min}}(\theta, \theta + h)$ is any tractable lower bound of the probability of error associated with the binary hypothesis test $\{H_0: \theta, H_1: \theta + h\}$ for equally likely hypotheses. The MSE bound is then obtained through the following equality [7]

$$E[\varepsilon^2] = \frac{1}{2} \int_{0}^{h} P_t\left(|\varepsilon| \geq \frac{h}{2}\right) dh.$$  

This scalar ZZB has been extended by Bell et al. to handle the $N$-dimensional vector parameter case [7]

$$a^T \Sigma a \geq \int_{0}^{h} V\left\{ \max_{\Delta \theta \in \Delta - h} \left[ \int_{R \times \min(p_o(\theta), p_o(\theta + \Delta))} P_{\text{min}}(\theta, \theta + \Delta) d\theta \right] dh, \right.$$  

where $\Sigma = E[\varepsilon \varepsilon^T]$ is the $N \times N$ error covariance matrix and $a$ is an arbitrary vector. When $a$ is the unit vector with a one in the $n$th position, (7) yields a bound on the MSE of the $n$th component of $\theta$. As can be seen in (7), the ZZB involves two integrations. Most of the time, and it is the case here, these integrals must be evaluated numerically. The computation of $P_{\text{min}}(0, \theta + \Delta)$ represents the central part of the ZZB derivation. In the absence of nuisance parameters in (3), i.e., when the source signal is perfectly known, this minimum error probability is given by the error probability of the Likelihood ratio test (LRT), which is optimal for testing $\{0, \theta_0\}$. However, the incoherent processor doesn’t have full knowledge of the source signal and therefore has to deal with nuisance parameters. Since the nuisance parameters are replaced by their ML estimates, the error probability of the Generalized Likelihood ratio test (GLRT) can be used as $P_{\text{min}}(0, \theta + \Delta)$ and can be computed using results on quadratic forms in Gaussian variables [8].

4. PERFORMANCE ANALYSIS

Performance analysis is performed in a simulated environment that mimics the New Jersey shelf during the Shallow Water 2006 (SW06) experiment. It is a range independent
waveguide that possesses a realistic sound speed profile derived from a conductivity-temperature-depth probe (CTD41) during the SW06 experiment. The water depth is $D = 79$ m. The seabed consists in two sediment layers overlying a semi-infinite basement. The thickness of the first layer is $D_1 = 5$ m, its sound speed and density are respectively $c_1 = 1650$ m/s and $\rho_1 = 1.7$ g/cm$^3$. The second layer thickness is $D_2 = 15$ m, its sound speed and density are respectively $c_2 = 1600$ m/s and $\rho_2 = 1.6$ g/cm$^3$. These two layers overlay a basement whose sound speed and density are $c_b = 1850$ m/s and $\rho_b = 1.9$ g/cm$^3$. We consider $K = 51$ frequencies regularly sampled in the frequency band $\Delta f = [50, 150]$ Hz and the first five propagating modes. The SNR is defined as $\gamma(f_k) = \frac{|s(f_k)|^2}{\sigma_w^2(f_k)} \| \mathbf{x}(f_k, \theta_0) \|^2$, and is the same for all frequencies.

The left panel of Fig. 1 shows the ZZB bound and Monte-Carlo simulations for range estimation. The source depth is fixed to $z_s = 35$ m and the range is unknown with a uniform prior distribution on the interval $r = [1000, 10000]$ m. The right panel of Fig. 1 shows the ZZB bound and Monte-Carlo simulations for depth estimation. The source range is fixed to $r_s = 5000$ m and the source depth is unknown with a uniform prior distribution on the interval $z = [1, 78]$ m. In both cases it can be seen that the ZZB closely follows the Monte-Carlo simulations which demonstrates the ability of the ZZB to predict MMP performance in all SNR regions. While range estimation clearly exhibits around 7 dB the characteristic threshold region (abrupt increase in MSE as SNR decreases) of non-linear estimation resulting from sidelobes on the ambiguity function, depth estimation does not show this phenomenon. In fact, it appears that depth estimation with known range does not suffer from ambiguity sidelobes.

![Fig 1: Performance bound for source range estimation (left panel) and for depth estimation (right panel). Solid line: ZZB; *: ML Monte-Carlo simulations.](image)

Performance analysis of MMP for joint estimation of range and depth are now carried through the ZZB alone as Monte-Carlo simulations become unpractical. The results are presented on Fig. 2. The left panel shows the ZZB bound on range and the right panel shows the ZZB bound on depth. The source range and depth are unknown with a uniform prior distribution on the interval $r = [1000, 10000]$ m and $z = [1, 78]$ m. Since range and depth estimates are coupled, the bound on range and the bound on depth both exhibit the threshold phenomenon around the SNR of 7 dB. In this threshold region, performance degrades significantly until it reaches the no-information region (region where the error no
longer changes) where performance is solely driven by the \textit{a priori} knowledge on range and depth.

![Performance bound for joint estimation of source range and depth, ZZB on range (left panel) and ZZB on depth (right panel).]

These kind of performance curves can be used to infer the necessary SNR that enables acceptable performance or to predict the achievable performance of well controlled experiments. Further analyses can consider mismatch or the coherent MMP processor.

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REFERENCES

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PERFORMANCE ANALYSIS OF ARCTIC TOMOGRAPHY
USING THE CRAMER RAO BOUND

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Abstract: We consider the minimum mean square error resolution limits on tomographic inversion for sound speed profile in the Arctic, using the Cramer Rao Bound (CRB) on parameters characterizing the halocline and pycnocline. The tomographic system for this example is a point source transmitting wide band coherent signals to a single vertical line array. The received signal model is the complex Gaussian gain (Swerling model II) multiplying the Green's functions from source to the receiver elements. The propagation model use a full wave field solution from OASES and an additive noise generated from a Kuperman-Ingenito surface wave model whose covariance is also generated by OASES. The CRB implements bounds for active, wide band signals so all the time resolved multipaths are incorporated as compared to previous narrowband passive analyses. We introduce this CRB approach as a paradigm for understanding the trade-offs among the several experimental design parameters such as source depth, receiver geometry and SNR tomographic inversions. While the problem considered here is a simple one, we suggest more sophisticated ones using range dependent propagation codes for the Green's functions and a computationally efficient evaluation of the CRB will lead to useful performance metrics complementing existing ones such as resolution kernels.

Keywords: Ocean tomography, Cramer Rao Bounds
1. INTRODUCTION AND BACKGROUND

Inversion of acoustic data to make tomographic inferences about the propagation medium has been the focus of many avenues of research. Ocean acoustic tomography, focalization, genetic algorithms and extension to particle filtering, Bayesian geoaoustic inversions, and several other algorithms are examples from the ocean acoustics literature. [1-4]. The literature for the theory and application on inversion of wave fields is huge and well beyond citing here. Most of the publications present an algorithm and show examples with either simulated or field data, or both. However, the accuracy of the estimates of the environmental parameters, or the resolution limits and the their coupling, have not been the dominant concern. Resolution kernels and posterior covariances are examples of current performance metrics, but they have in general not been coupled to physics-based propagation and ambient noise models. This motivates the topic of this paper.

We use an efficient formulation of the Cramer Rao Bound (CRB) coupled to a high-fidelity, wave theory propagation model, OASES[5], as well as the source waveforms and receiver arrays to establish lower bounds of the mean square error of the parameter estimates, independent of any algorithm used. The significance of this approach is that it couples the propagation physics, the source/receiver properties and performance limits with the signal processing, into one consistent performance analysis framework. We emphasize this is a broadband model, so the ray/mode structure as well as arrival angles across an array are implicitly included.

The motivation for this research has been recent scientific initiatives for environmental assessment in the Arctic using Ocean Acoustic Tomography (OAT), sponsored by various agencies, including ONR. Acoustic propagation in the Arctic is unique since the sound speed channel minimum is at the surface and is tightly coupled by the strong gradients of the halocline and pycnocline, so the acoustic propagation is very different from that experienced in more temperate oceans. Consequently, the selection of tomographic system parameters such as source depth, receiver array geometry, waveforms and source/receiver separation and their effect on the tomographic inversions accuracy represents new scientific territory, and is complicated at best.

2. ARCTIC PROPAGATION AND TOMOGRAPHY

Aside from being a half channel the Arctic has two major features: i) a halocline and ii) a pycnocline. These are illustrated in Fig. 1.
Fig. 1. Sound speed profiles representative of the deep Arctic: Left - SVP to 4000m indicating pycnocline and start of the hydrostatic gradient, Right - Expanded SVP to 400 m indicating halocline and pycnocline

The halocline is a layer typically 50 – 60 m deep caused by the precipitation from "salt fingers" in the ice, so starting at approximately 30 Hz modes can be ducted in this channel. These modes are very time dispersive. In addition, when an ice cover is present, the scattering by roughness and absorption by conversion to shear, leads to abnormally high loss for these surface trapped modes [6]. These shallow modes or their ray equivalents are only observed by sensors in the surface duct. Below the halocline, the pycnocline layer represents another important environmental acoustics feature, a 200 – 250 meter deep layer which separates the cold surface from the deeper isothermal water. This feature forms another surface channel with convergence zone features, resulting in discrete ranges with lower transmission loss (TL). Only the sensors which are above the pycnocline receive these modes/rays. There is also a deeper layer of nominally 1 km thickness, above the depth where the SVP is dominated by the hydrostatic gradient common to all oceans. This transition is caused by the interplay of weak diffusion of colder and denser water at depth and the hydrostatic pressure.

3. CRAMER RAO BOUND FORMULAE

In the analysis we use a Gaussian model for the signal and noise. The following describes the signals generated by a wide band source and observed at a receiver, here a VLA:

$$\mathbf{r}^l = \mathbf{G}(\mathbf{a})\mathbf{b}^l + \mathbf{n}^l$$  \hspace{1cm} (1)$$

where $\mathbf{G}(\mathbf{a})$ is the Green's function from a source to a receiver, $\mathbf{b}$ is a complex Gaussian random vector with covariance $\mathbf{K}_b$ which scales with the source signals, $\mathbf{n}$ is an additive complex Gaussian noise with covariance $\mathbf{K}_n$, and $l=1,L$ with $L$ being the number of observations, or 'snapshots'.

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This is a very general model. The observation vector can be time samples or frequency domain samples of an STFT at a sensor array with each having vector and/or polarization components. With a frequency domain representation for example the overall dimension is the product of frequency bins, sensors and components is \( N = N_f N_a N_v \), which can be a very large number.

Parameter estimation and performance bounds have been the subject of a significant amount of work reported in the signal processing literature [7]. The signal processing literature divides estimation accuracy into three regions usually according to SNR. We have: i) At high SNR's the estimator is operating in what is called the linear region. This is where methods based on local linearity and Taylor expansions model the signal processing well. Two important issues here are bias "model mismatch". For the former errors occur because a model may be close, but not exact, so a higher SNR has less value or saturates. ii) At medium SNR's there is a transition to where an estimators is sometimes picking false, or ambiguous, peaks in the matched filtering operation implicit in most parameter estimators. The important issue here is the threshold at which the performance enters this region. iii) Finally, if the SNR is so low that there is no mutual information between source and receiver, one must rely on \emph{a priori} information.

The CRB establishes a lower bound on the MMSE of an unbiased estimate of a parameter. The bound is valid everywhere, but most useful for the linear region where estimation is useful. It is used extensively in sonar and radar signal processing to benchmark the performance of estimators for source localization, e.g. range, bearing and depth, but is less so used for tomographic estimation. This is the application considered here.

We summarize the CRB formulation from previous results [8] which have been recently extended to the much more general case when \( \mathbf{b} \) is a vector with correlation among the elements. The CRB is determined by the Fisher Information Matrix (FIM). We specialize the general case to when \( \mathbf{b} \) is a scalar given by \( S_s(f) \mathbf{b} \) as done in [8] where \( S_s(f) \) is the energy spectrum of the transmitted signal. For this define the following four quantities as specified by the signal model:

\[
\begin{align*}
    d^2(\mathbf{a}) &= \int_{f_a - W/2}^{f_a + W/2} \left| S_s(f) \right|^2 \sigma_b^2 G(f, \mathbf{a}) K_n^{-1}(f) G^T(f, \mathbf{a}) df \\
    I_i(\mathbf{a}) &= \int_{f_a - W/2}^{f_a + W/2} \left| S_s(f) \right|^2 \sigma_b^2 \frac{\partial^2 G(f, \mathbf{a})}{\partial a_i} K_n^{-1}(f) G^T(f, \mathbf{a}) df \\
    I_{ij}(\mathbf{a}) &= \int_{f_a - W/2}^{f_a + W/2} \left| S_s(f) \right|^2 \sigma_b^2 \frac{\partial^2 G(f, \mathbf{a})}{\partial a_i \partial a_j} K_n^{-1}(f) G^T(f, \mathbf{a}) df \\
    \gamma(\mathbf{a}) &= \left[ 1 + d^2(\mathbf{a}) \right]^{-1}
\end{align*}
\]

The diagonal of the inverse of the FIM specifies a lower bound on any unbiased estimate of a parameter while the off diagonal terms indicate the coupling among the parameters. The elements of the FIM, \( J_{ij} \), are given by
We note that this is the formulation of the CRB for "active" acoustic tomography problem. There a few earlier publications on using this CRB formulation for tomographic applications. Reference [9] applies them for a narrowband, passive model for seabed inversion while [10] compares active and passive acoustics in the context of a very simplified model for global warming.

4. APPLICATION TO ARCTIC TOMOGRAPHY

As an example we consider the problem of estimating the depths of the halocline and the pycnocline indicated in Fig. 1. The oceanographic parameters of principal interest in regard to OAT are those characterizing the shallow depths most sensitive to climate change. Therefore, the obvious parameters of interest in the Arctic are the depths of the halocline and pycnocline responsible for the dominance of surface duct propagation. In temperate oceans the important ray paths are the deep diving ones which have turning points at the depths near the surface. In contrast, in the upward refracting Arctic environment, the deep diving rays are propagating at relative steep angles near the surface and do not have any turning points where the propagation becomes horizontal, and these propagation path are therefore less sensitive to climate induced changes in the halo- and pycnoclines. The sensitive rays are the shallow ones corresponding to the low order modes trapped in the surface channel, as was observed in the TAP experiment [11]. To understand this fundamentally different physics associated with tomographic inversion in the Arctic, we examine the transmission loss and the time series of a vertical line array receiver.

As an example, Fig. 2 shows the transmission loss at 250 Hz for ranges out to 500 km for a source at depth 100 m, clearly illustrating the dominance of the surface ducting, here controlled by the pycnocline in particular. The propagation is strongly ducted with nearly cylindrical spreading above the pycnocline depth at 250 m. There is also a weak ducting above 1000 m depth, associated with the switch to the lower hydrostatic gradient. Finally, there is the typical CZ cycling at range intervals of approximately 60 km associated with the hydrostatic gradient. The contribution of its periodic focusing in range in the surface duct is low compared to pycnocline ducting, in contrast to the temperate ocean where the CZ path are the dominant features for the acoustic propagation from shallow sources.

\[
\begin{align*}
J_{i,j}^1(a) &= 2L \left[ J_{i,j}^1(a) + 2J_{i,j}^2(a) \right] \\
J_{i,j}^2(a) &= \gamma(a) \text{Re} \left( d^2(a) l_i^j(a) + l_i^j(a) \gamma(a) \right) \\
J_{i,j}^3(a) &= \gamma(a) \text{Re} \left( l_i^j(a) \right) \text{Re} \left( l_i^j(a) \gamma(a) \right)
\end{align*}
\]
Fig. 2. Narrow band transmission loss in an Arctic environment with SVP shown in Fig. 1, and a source at depth 100 m.

Fig. 3. Time series modeling for SVP in Fig. 1, and broadband source at 100 m depth. Left: depth stack at 100 km range. Right: range stack at 250 m depth, ranges 20-200 km, Stacking reduction velocity 1.485 m/s.

The computed TL in Fig. 2 affirms that at even modest ranges a VLA receiver is best concentrated above the pycnocline layer and deep and large apertures are not needed to intercept the most energetic rays/modes. The significance of this observation to the design of tomographic networks in the Arctic is obvious.

For the same range-independent SVP, Fig. 3 illustrates the timeseries of the received signals vs. depth and range for wide band, 200 – 250 Hz, signal. The received signals are plotted using a stacking reduction velocity (moving time window) of 1485 m/s. The early arrivals of the depth stack at 100 km in the left frame indicates the deep diving, but relatively fast, rays/modes with their two surface bounces, one near the source, the other near the receiver. The latest arrivals correspond to the low order modes trapped by the pycnocline, with the amplitudes decreasing with depth, consistent with the TL in Fig. 2. At least three modes can be discerned. The early arrivals on the range stack at 250 m depth are again the deep diving rays/modes. These rays/modes go deeper with an increasing range.
which leads to the "arc" shape. The reason for this phenomenon is that these rays turn at a higher phase speed and consequently group speed because of the positive dispersion. The approximately 50-60 km convergence zone distance is evident from the range separation of the early “ray” arrivals. Also, the range stack shows the modal spreading separation with increasing range, associated with the positive group speed dispersion characteristic of the Arctic environment.

Fig. 4. Cramer-Rao resolution vectors for estimating the depths of the halo- and pycnocline through tomographic inversion of 200-300 Hz wideband signal transmitted by a source at 50 m depth and received by a vertical line array at 250 km distance. Left: Mean array depth 30 m. Right: Mean array depth 130 m.

To illustrate the significance of the source/receiver configuration to the performance of tomographic inversion in the Arctic, Fig. 4 shows the Cramer-Rao resolution vectors for the estimation of the depth of the halo- and pycnocline. The computation is performed using the OASES modeling infrastructure originally developed for geo-acoustic inversion [9]. A wide band acoustic source, deployed at the halo-pycnocline transition at depth 50 m, is transmitting a 200-300 Hz signal at a spectral level of 130dB. The signal is received on a vertical array at 250 km range, deployed at two depths, one in the halocline at depth 30 m (left frame), the other in the pycnocline at 130m depth (right frame). In both cases the array has 16 elements, spaced at 3 m. The plots show the principal axes of the CRB matrix, representing the one standard deviation minimum uncertainty of the depth resolutions for the halo- and pycnocline estimation and their coupling. The most noteworthy feature is the extremely high resolution of the halocline depth, two orders of magnitude better than for the pycnocline depth. This feature suggests that the dominant feature is the gradient in the two layers. Also, the results show the improved performance for both estimates when the array is deployed in the halocline, exploiting virtually all propagation paths. Finally, there is little coupling between the estimate uncertainties, in particular for the shallow array.

5. CONCLUSIONS

A comprehensive environmental acoustic modeling infrastructure has been combined with the Cramer-Rao lower bound formulation to provide an efficient computational tool for
evaluating the performance of tomographic networks for assessing the oceanographic signatures of climate change in the Arctic. The analysis has shown several fundamental differences in the optimal design of the tomographic network compared to the traditional OAT at more temperate latitudes, with the most significant being the reduced role of the deep diving ray paths for the inversion, and the dominant role of the late arriving modal field in the surface channel, suggesting a transition to Matched Field Tomography exploiting the relative phases across the array and reducing the use of absolute ray travel times as basis for the tomographic inversion.

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SOURCE MOTION PARAMETER ESTIMATION USING DIRECT AND MULTIPATH ARRIVALS AT A PAIR OF HYDROPHONES

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Abstract: The signal emitted by an acoustic source moving on the sea surface or underwater arrives at a hydrophone located above the sea floor via a direct path and one or more multipaths. By exploiting the direct and multipath arrivals at a pair of hydrophones, it is possible to estimate the complete set of five motion parameters of the source as it travels past the hydrophone pair in a straight line at constant speed and constant height above the sea floor, provided that it is known a priori on which side of the hydrophone pair the source’s closest point of approach is located. When the source is a surface vessel and the two hydrophones are positioned close to the sea floor in a shallow water environment, the cross-correlogram of the hydrophone pair shows three dominant, distinct tracks. The middle track, with positive amplitude, represents the temporal variation of the differential time of arrival (DTOA) between the direct path signals at the two hydrophones. The upper or lower track, with negative amplitude, represents the temporal variation of the DTOA between the direct path signal at one hydrophone and the bottom-surface-reflected path signal at the other hydrophone. A model for these three tracks is derived, which is a function of time and the five motion parameters of the source. A least-squares fit of this model to the observed tracks provides estimates of all five motion parameters of the surface vessel. The effectiveness of this source motion parameter estimation method is demonstrated using real data recorded from a bottom-mounted hydrophone array in a shallow water experiment for the transits of two different vessels.

Keywords: Motion parameter estimation, hydrophone pair, acoustic source, multipath arrival, differential time of arrival
1. INTRODUCTION

The trajectory of an acoustic source moving along a linear path on the sea surface or under water at constant speed and constant depth is fully specified by a set of five motion parameters [1]. The signal emitted by the acoustic source arrives at a hydrophone located above the sea floor via a direct path and one or more multipaths. A bottom-mounted wide aperture array consisting of multiple, widely separated hydrophones can be used to estimate all five motion parameters of the source as it transits past the array. The basic principle of the source motion parameter estimation is to measure the temporal variation of the differential time of arrival (DTOA), or the time delay, of the direct path signal at each sensor pair of the array and then to minimize the sum of the squared deviations of the time delay estimates from their predicted values over a sufficiently long period of time for all sensor pairs [1, 2]. However, with this method, the source motion parameters are \textit{unidentifiable} when the array consists of only a pair of hydrophones [3]. In this case, reliable estimates of the source motion parameters may be obtained by exploiting the direct and multipath arrivals at the hydrophone pair, provided it is known \textit{a priori} on which side of the hydrophone pair the source’s closest point of approach is located. A similar approach was adopted previously to estimate the full set of flight parameters of a jet aircraft using the direct and ground-reflected path arrivals at a small aperture microphone array [4]. In this paper, the direct and bottom-surface-reflected path arrivals at a pair of hydrophones positioned close to the sea floor in a shallow water environment are used to estimate the five motion parameters of a surface vessel as it transits past the hydrophone pair. The source motion parameter method is formulated and its effectiveness demonstrated using real data recorded from a bottom-mounted hydrophone array in a shallow water experiment for the transits of two different vessels.

2. SOURCE MOTION MODEL

Figure 1 shows the geometrical configuration for an array of two hydrophones located above a flat sea floor and the linear trajectory of an underwater acoustic source as it transits past the array in a straight line at constant speed and constant depth below the sea surface. The water depth is \( w \). The \( xy \)-plane coincides with the sea floor and the positions of sensors 1 \((S_1)\) and 2 \((S_2)\) are given by \((0,0,z_1)\) and \((d,0,z_2)\) respectively. The source position at any time \( t \) is given by

\[
\begin{align*}
x(t) &= d_e \cos \theta_c + (t - \tau_c)V \sin \theta_c \\
y(t) &= d_e \sin \theta_c - (t - \tau_c)V \cos \theta_c \\
z(t) &= h
\end{align*}
\]

where \( V \) is the source velocity, \( \tau_c \) is the time when the source is at the closest point of approach (CPA) to the origin O, \( h \) is the source height (above the sea floor), \( d_e \) is the source horizontal range at CPA, and \( \theta_c \) \((-\pi < \theta_c < \pi\) is the source azimuth angle at CPA. The source trajectory is completely specified by the five motion parameters \( \{V, \tau_c, h, d_e, \theta_c\} \). Note that the source velocity \( V \) is either positive or negative, depending on whether the origin O is on the source’s right or left hand side as the source moves along its trajectory. Also, the source speed \( |V| \) is much less than the speed \( c \) of sound.
propagation in water so that the retardation effect (i.e., the displacement of the source during the propagation of the emitted signal to the sensors) can be ignored.

Fig. 1. Geometrical configuration for a hydrophone pair ($S_1$ and $S_2$) located above a flat sea floor (xy-plane) and the linear trajectory of an underwater acoustic source.

Fig. 2. Direct and bottom-surface reflected paths for the signal emitted by the source at time $t$ arriving at $S_n$, and the mirror image ($S'_n$) of $S_n$ for the bottom-surface reflection.

3. TIME DELAY AND INTERSENSOR MULTIPATH DELAYS

The source emits continuously a broadband acoustic signal, which arrives at each hydrophone via a direct path and one or more multipaths. Figure 2 shows the direct and bottom-surface-reflected paths for the signal emitted by the source at time $t$ arriving at $S_n$ ($n = 1, 2$), and the mirror image ($S'_n$) of $S_n$ for the bottom-surface reflection. Note that the $x$-coordinate of $S_n$, denoted as $x_n$ in Fig. 2, is equal to zero for $n = 1$, and $d$ for $n = 2$. Let
\[ D_{21}^{dd}(t) \] denote the DTOA between the direct path signal at \( S_2 \) and the direct path signal at \( S_1 \) at time \( t \), \( D_{21}^{lr}(t) \) the DTOA between the bottom-surface-reflected path signal at \( S_2 \) and the direct path signal at \( S_1 \) at time \( t \), and \( D_{21}^{dl}(t) \) the DTOA between the direct path signal at \( S_2 \) and the bottom-surface-reflected path signal at \( S_1 \) at time \( t \). By definition,

\[
D_{21}^{\alpha\beta}(t) = \left[ R_n^{\alpha}(t) - R_n^{\beta}(t) \right]/c \quad \text{for } \{\alpha, \beta\} = \{d, d\}, \{r, d\} \text{ and } \{d, r\}
\]

where \( R_n^{\alpha}(t) \) and \( R_n^{\beta}(t) \) are the lengths of the direct path and bottom-surface-reflected path from the source to \( S_n \) at time \( t \), respectively.

The DTOA between the direct path or a multipath signal at one sensor and the direct path or a multipath signal at the other sensor can be estimated by first cross-correlating the outputs of the two sensors and then finding the time lag corresponding to the appropriate peak of the cross-correlation function. When the source moves on (or just below) the sea surface and the two hydrophones are located close to the sea floor, the cross-correlation function for \( S_1 \) and \( S_2 \) at any time \( t \) is dominated by a strong positive peak and two weaker negative peaks. The positive peak corresponds to the cross-correlation peak of the direct path signal components at the outputs of the two sensors, and each of the two negative peaks corresponds to the cross-correlation peak of the direct path signal component at the output of one sensor and the bottom-surface-reflected path signal component at the output of the other sensor (noting that the signal is inverted after it is reflected by the sea surface). The time lag corresponding to the positive peak provides an estimate of \( D_{21}^{dd}(t) \), while the time lags corresponding to the two negative peaks provide estimates of \( D_{21}^{rd}(t) \) and \( D_{21}^{dr}(t) \) respectively. In this paper, \( D_{21}^{dd}(t) \) is referred to as time delay, while \( D_{21}^{rd}(t) \) and \( D_{21}^{dr}(t) \) are both referred to as intersensor multipath delays.

From Fig. 2, \( R_n^{d}(t) \) and \( R_n^{r}(t) \) can be written, respectively, as

\[
R_n^{d}(t) = \left\{ [x(t) - x_n]^2 + y^2(t) + (h - z_n)^2 \right\}^{1/2}
\]
\[
R_n^{r}(t) = \left\{ [x(t) - x_n]^2 + y^2(t) + (h - z_n + 2w)^2 \right\}^{1/2}.
\]

Substituting (1) into (3) and (4), and after some manipulation, gives

\[
R_n^{d}(t) = \left\{ V^2 (t - \tau_c)^2 + d_c^2 + (h - z_n)^2 - 2x_n [d_c \cos \theta_c + (t - \tau_c)V \sin \theta_c] + x_n^2 \right\}^{1/2}
\]
\[
R_n^{r}(t) = \left\{ V^2 (t - \tau_c)^2 + d_c^2 + (h - z_n + 2w)^2 - 2x_n [d_c \cos \theta_c + (t - \tau_c)V \sin \theta_c] + x_n^2 \right\}^{1/2}
\]

for \( n = 1, 2 \). Equations (2), (5) and (6) constitute a delay model that predicts the temporal variations of the time delay and intersensor multipath delays for the sensor pair \( S_1 \) and \( S_2 \). Defining the source motion parameter vector \( \lambda = [V, \tau_c, h, d_c, \theta_c]^T \), this delay model is a function of time \( t \) and \( \lambda \), i.e., \( D_{21}^{\alpha\beta}(t, \lambda) \) for \( \alpha \beta = dd, rd \) and \( dr \).

### 4. SOURCE MOTION PARAMETER ESTIMATION

Let \( \hat{D}_{21}^{\alpha\beta}(t) \) be the estimate of \( D_{21}^{\alpha\beta}(t) \). The following nonlinear least-squares (NLS) solution is proposed as an estimate of \( \lambda \):
where \( \lambda' = [V', \tau_c', h', d_c', \theta_c']^T \), \( \hat{\lambda} = [\hat{V}, \hat{\tau}_c, \hat{h}, \hat{d}_c, \hat{\theta}_c]^T \), \( \hat{f}(t_i) = [\hat{D}^{dd}_{21}(t_i), \hat{D}^{rd}_{21}(t_i), \hat{D}^{dr}_{21}(t_i)]^T \) and \( f(t_i, \lambda') = [D^{dd}_{21}(t_i, \lambda'), D^{rd}_{21}(t_i, \lambda'), D^{dr}_{21}(t_i, \lambda')]^T \) are the observation and model vectors at time \( t_i \) (\( 1 \leq i \leq M \)), respectively, and \( \| \cdot \| \) denotes the \( l_2 \) norm of a vector. The minimization is implemented in MATLAB using the global optimization solver GlobalSearch in conjunction with the minimization algorithm \textit{fmincon}, subject to the following set of constraints (lower and upper bounds) on the motion parameters:

\[
\begin{align*}
\frac{1}{2} (\text{sgn}(V) - 1)c < V' < \frac{1}{2} (\text{sgn}(V) + 1)c, & \quad \hat{\tau}_c^o - \Delta \tau_{\text{max}} < \tau_c' < \hat{\tau}_c^o + \Delta \tau_{\text{max}} \\
0 < h' < w, & \quad 0 < d_c' < d_{\text{c,max}}, \quad \frac{1}{2} (\text{sgn}(\theta_c) - 1)\pi < \theta_c' < \frac{1}{2} (\text{sgn}(\theta_c) + 1)\pi
\end{align*}
\]

where \( \hat{\tau}_c^o \) is the initial estimate of \( \tau_c \), \( \Delta \tau_{\text{max}} \) is the maximum possible error in \( \hat{\tau}_c' \), \( d_{\text{c,max}} \) is the maximum CPA horizontal range of the source, and \( \text{sgn}(\cdot) \) denotes the sign of the quantity in brackets. In this paper, \( \Delta \tau_{\text{max}} = 10 \text{ s} \), \( d_{\text{c,max}} = 200 \text{ m} \), and the left-right ambiguity problem is resolved by assuming \textit{a priori} knowledge of \( \text{sgn}(\theta_c) \) (which is equivalent to knowing on which side (left or right) of the array the source’s CPA position is located). The sign of \( V \) is determined using the time delay measurements: when \( \theta_c > 0 \), \( V \) is negative if \( \hat{D}^{dd}_{21}(t_M) > \hat{D}^{dd}_{21}(t_1) \), and positive otherwise; whereas, when \( \theta_c < 0 \), \( V \) is negative if \( \hat{D}^{dd}_{21}(t_M) < \hat{D}^{dd}_{21}(t_1) \), and positive otherwise. A set of initial estimates for the motion parameters is required to start the numerical optimization. An initial estimate of \( \tau_c \) is obtained by finding the time when the received signal energy at \( S_1 \) is a maximum. The initial estimate of \( \theta_c \) is set to \( 0^\circ \). The initial estimates of \( |V| \) and \( d_c \) are assigned some typical values: 5 m/s and 50 m respectively. The water depth \( w \) is used as an initial estimate of \( h \).

\[\text{Fig. 3. The XY-coordinates of the eight hydrophones (H1 to H8) and the nominal trajectories of the small vessel (first scenario) and RHIB (second scenario).}\]
5. EXPERIMENTAL RESULTS

In a shallow water experiment (water depth \( \approx 21.5 \) m), eight hydrophones (labelled \( H_1 \) to \( H_8 \)) were located at a nominal height of 1 m above the sea floor. Figure 3 shows the sensor configuration, which was (almost) linear with an intersensor spacing of about 14 m. Acoustic data were recorded from the bottom-mounted hydrophone array for two different scenarios: a small vessel transit and a rigid-hulled inflatable boat (RHIB) transit, and their nominal trajectories are shown respectively as blue and red lines in Fig. 3. The nominal values of the source motion parameters for both transits are shown in Table 1. The output of each hydrophone was sampled at a frequency of 250 kHz. The eight hydrophones could be used to form seven sensor pairs, each consisting of two adjacent hydrophones. However, due to data acquisition problems among some hydrophones, only four sensor pairs were formed: \( \{S_1, S_2\} = \{H_3, H_4\}, \{H_5, H_6\}, \{H_6, H_7\}, \) and \( \{H_7, H_8\} \).

(a) First scenario: small vessel transit

<table>
<thead>
<tr>
<th>Parameter</th>
<th>( V ) (m/s)</th>
<th>( \tau_c ) (s)</th>
<th>( h ) (m)</th>
<th>( d_c ) (m)</th>
<th>( \theta_c ) (deg)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Nominal value</td>
<td>-4.6</td>
<td>2665.5</td>
<td>21.5</td>
<td>135.3</td>
<td>16.8</td>
</tr>
<tr>
<td>Mean value</td>
<td>-4.4</td>
<td>2665.8</td>
<td>20.6</td>
<td>132.2</td>
<td>17.2</td>
</tr>
<tr>
<td>Standard deviation</td>
<td>0.1</td>
<td>0.3</td>
<td>1.1</td>
<td>1.5</td>
<td>0.3</td>
</tr>
</tbody>
</table>

(b) Second scenario: RHIB transit

<table>
<thead>
<tr>
<th>Parameter</th>
<th>( V ) (m/s)</th>
<th>( \tau_c ) (s)</th>
<th>( h ) (m)</th>
<th>( d_c ) (m)</th>
<th>( \theta_c ) (deg)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Nominal value</td>
<td>10.8</td>
<td>81.4</td>
<td>21.5</td>
<td>42.8</td>
<td>93.1</td>
</tr>
<tr>
<td>Mean value</td>
<td>10.2</td>
<td>81.1</td>
<td>20.1</td>
<td>41.1</td>
<td>93.6</td>
</tr>
<tr>
<td>Standard deviation</td>
<td>0.2</td>
<td>0.1</td>
<td>0.9</td>
<td>0.8</td>
<td>1.3</td>
</tr>
</tbody>
</table>

Table 1: Mean values and standard deviations in the estimates of the source motion parameters and their nominal values for (a) first scenario and (b) second scenario.

For a given sensor pair, the data from each sensor were processed in blocks, each containing 65,536 samples, with 50% overlap between two consecutive blocks. Each data block from one sensor was cross-correlated with the corresponding data block from the other sensor. The phase transform prefiltering technique was used to suppress ambiguous peaks which would otherwise have appeared in the cross-correlation function due to the strong harmonic components of the source signal. The generalized cross-correlation processing (with phase transform prefiltering) was implemented in the frequency domain using the fast Fourier transform, with a spectral window from 50 to 1000 Hz. In this way, a cross-correlogram was produced for each hydrophone pair, which is an amplitude plot showing the temporal variation of the cross-correlation function, with the horizontal axis corresponding to time and vertical axis corresponding to time lag variable of the cross-correlation function. Figures 4 (a) and 5(a) show the cross-correlograms of the second sensor pair: \( \{S_1, S_2\} = \{H_3, H_4\} \) for the first and second scenarios respectively. In both figures, three time-lag tracks can be observed, with the upper (negative amplitude), middle (positive amplitude), and lower (negative amplitude) tracks representing \( \hat{D}_{21}^{cl}(t), \hat{D}_{21}^{dl}(t), \) and \( \hat{D}_{21}^{dr}(t) \), respectively. Similar observations can be made for all other sensor pairs.
Fig. 4. Results for the small vessel transit. (a) Cross-correlogram of the second sensor pair \{H_5, H_6\}, and (b) time delay and intersensor multipath delay estimates extracted from (a) together with the LS fit of the delay model to these delay estimates.

Fig. 5. Results for the RHIB transit. (a) Cross-correlogram of the second sensor pair \{H_5, H_6\}, and (b) time delay and intersensor multipath delay estimates extracted from (a) together with the LS fit of the delay model to these delay estimates.

The proposed source motion parameter estimation method was applied in turn to each of the four sensor pairs. Figure 4(b) shows the sequence of time delay estimates and the two sequences of intersensor multipath delay estimates from the second sensor pair for the first scenario (represented by small circles), together with the least-square (LS) fit of the delay model to these time sequences of delay measurements (represented by solid lines). Figure 5(b) shows the corresponding result for the second scenario. Note that in processing the data for each sensor pair, a local $xy$-coordinate system was set up on the sea floor so that the projections of the first and second sensors ($S_1$ and $S_2$) onto the $xy$-plane were located at the origin and on the positive $x$-axis respectively. Thus, for the same vessel transit, the CPA time and CPA horizontal range (which were measured with respect to the origin of the local coordinate system) for one sensor pair were different from those for another sensor pair. Also, the orientation of the $x$-axis of the local coordinate system was
(slightly) different for each sensor pair (as the configuration for the eight hydrophones was not perfectly linear), so the CPA azimuth angle (which was measured with respect to the x-axis) for one sensor pair was (slightly) different from that for another sensor pair. To ensure that each sensor pair provided an estimate of the same set of source motion parameters, the CPA azimuth angle estimate from each sensor pair was corrected so that they were all measured with respect to the X-axis of the global coordinate system as shown in Fig. 3. Also, the CPA time and CPA horizontal range estimates from each sensor pair were corrected so that they were all measured with respect to $H_1$. These corrections were done using the surveyed positions (global coordinates) of the eight hydrophones. The statistics of the four data sets of motion parameter estimates (one data set per sensor pair) were then computed, and the results for both scenarios are shown in Table 1.

6. CONCLUSIONS

A method has been described which uses both the direct and bottom-surface-reflected path arrivals at a pair of hydrophones positioned close to the sea floor to estimate all five motion parameters of a transiting acoustic source moving on (or just below) the sea surface in a shallow water environment. Experimental results have demonstrated the effectiveness of the proposed source motion parameter estimation method for two different sources: a small vessel and a RHIB, using a pair of hydrophones that are 14 m apart and located 1 m above the sea floor in 21.5 m of water. A Cramer-Rao lower bound (CRLB) error analysis of the method (not presented in this paper) has also indicated its potential of providing reliable source motion parameter estimates using a pair of closely spaced hydrophones.

REFERENCES

SPARSITY AND SUPER-RESOLUTION IN SOUND SOURCE LOCALIZATION WITH SENSOR ARRAYS

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\textbf{Abstract}: Sound source localization with sensor arrays involves the estimation of the direction-of-arrival (DOA) from a limited number of observations. Compressive sensing (CS) is a method for solving such undetermined problems which achieves simultaneously sparsity, thus super-resolution, and computational speed. We formulate the DOA estimation as a sparse signal reconstruction problem and show that methods which exploit sparsity have superior performance compared to traditional methods for DOA estimation. To demonstrate the high-resolution capabilities and the robustness of CS and other sparsity promoting optimization techniques in DOA estimation, the methods are applied to experimental data from underwater acoustic measurements in the challenging scenario of source tracking from single snapshot data.

\textbf{Keywords}: Sparsity, compressive sensing, direction of arrival (DOA) estimation, sensor arrays
1. INTRODUCTION

The problem of direction-of-arrival (DOA) estimation with sensor arrays is to infer the number and the location of (usually few) sound sources possibly in the presence of noise from measurements of the wavefield with an array of sensors. Conventional beamforming [1] is the simplest traditional method for DOA estimation, though it is characterized by low resolution. Other methods [1], developed to overcome the resolution limit of conventional beamforming, have degraded performance under noisy conditions, coherent sources and sample-starved data.

The compressive sensing (CS) framework [2] asserts that signals can be reconstructed from very few measurements as long as the signals are sparse and can be done with computationally efficient methods by solving a convex minimization problem with linear programming. CS and other sparsity promoting methods outperform traditional methods which aim to minimize the energy of the signal resulting in low-resolution, non-sparse solutions.

2. SPARSITY AND COMPRESSIVE SENSING

For simplicity, we formulate the DOA estimation problem assuming that the sources are in the far field of the array (i.e. plane waves), the processing is narrowband and the problem is confined in two dimensions (2D) with a linear array of sensors with known geometry.

Let $x \in \mathbb{C}^N$ be an unknown vector comprising the source strengths at all directions $\theta$ relative to the array axis on the angular grid of interest. Usually, there are only few sources $K \ll N$ present, resulting in a sparse $x$. Let $y \in \mathbb{C}^M$ be the vector of the wavefield measurements at the $M$ sensors linearly related to the signal $x$, such that in the absence of noise,

$$ y = A_{M \times N} x. \quad (1) $$

The sensing matrix $A$ is formed by concatenating the steering vectors,

$$ a(\theta_k) = \frac{1}{\sqrt{M}} e^{j \frac{2\pi}{\lambda} r_k \sin \theta_k}, $$

at all potential source directions $A = [a(\theta_1), ..., a(\theta_N)]$, where $\lambda$ is the wavelength and $r = [r_1, ..., r_M]^T$ the sensor locations.

Practically, we are interested in a fine resolution on the angle grid, thus $M < N$ and the problem (1) is underdetermined. A way of solving this ill-posed problem is to constrain the possible solutions with prior information.

Traditional methods impose a minimum $l_2$-norm constraint on the solution and solve (1) through the minimization problem,

$$ \min_{x \in \mathbb{C}^N} \|x\|_2 \quad \text{subject to} \quad y = Ax. \quad (P_2) $$
The convex problem \((P_2)\) aims to minimize the energy of the signal rather than its sparsity, resulting in a non-sparse solution, \(\hat{x} = A^H \left( AA^H \right)^{-1} y\). Beamforming is based on the \(l_2\)-norm method with the simplifying assumption, \(AA^H = I_M\), such that, \(\hat{x} = A^H y\).

By definition, sparsity can be imposed on the vector \(x\) by minimizing the \(l_0\)-norm
\[
\left\| x \right\|_0 = \sum_{i=1}^{N} \mathbb{I}_{x_i \neq 0},
\]
which counts the number of non-zero entries in the vector, leading to the minimization problem \((P_0)\),
\[
\min_{x \in C^N} \left\| x \right\|_0 \quad \text{subject to} \quad y = Ax.
\]

However, the minimization problem \((P_0)\) is a non-convex combinatorial problem, which becomes computationally intractable even for moderate dimensions. The breakthrough of CS came with the proof that for sufficiently sparse signals and sensing matrices with sufficiently incoherent columns the \((P_0)\) problem is equivalent to the \((P_1)\) problem [3],
\[
\min_{x \in C^N} \left\| x \right\|_1 \quad \text{subject to} \quad y = Ax,
\]
where the \(l_0\)-norm is replaced with the \(l_1\)-norm. The \(l_1\) relaxation \((P_1)\) of the \((P_0)\) problem is the closest convex optimization problem to \((P_0)\) and can be solved efficiently with linear programming [4] even for large dimensions.

For noisy measurements, \(y = Ax + n\), where \(n \in C^M\) is additive noise with bounded norm \(\left\| n \right\|_2 \leq \varepsilon\), the \((P_1)\) problem is reformulated as [5],
\[
\min_{x \in C^N} \left\| x \right\|_1 \quad \text{subject to} \quad \left\| Ax - y \right\|_2 \leq \varepsilon.
\]

The solution to \((P_1^\varepsilon)\) has the minimum \(l_1\)-norm while it fits the data up to the noise level.

CS offers super-resolution due to the sparsity constraint and computational efficiency due to convex relaxation of the \(l_0\)-norm optimization problem. However, as all DOA estimation methods, it has also limitations. A performance analysis of CS in DOA estimation, in terms of the discretization of the angular space, the coherence of the sensing matrix and the signal to noise ratio (SNR), is carried out in [6].

3. ENHANCING SPARSITY

The \(l_1\)-norm of a vector \(x\) is a convex function hence the optimization problem \((P_1^\varepsilon)\) converges to a global minimum. However, the solution to \((P_1^\varepsilon)\) is not necessarily the sparsest feasible. To enhance sparsity, the \(l_1\)-norm of the vector \(x \in C^N\) can be
replaced by other sparsity promoting functions such as the $l_p$-norm, 
\[
\|x\|_p = \left( \sum_{i=1}^{N} |x_i|^p \right)^{1/p}, \text{ with } 0<p<1 \ [7]; \text{ see Fig.1.}
\]

![Geometric visualization of (a) the $l_2$-norm, (b) the $l_1$-norm and (c) the $l_p$-norm problem with $0<p<1$ in $R^2$. The solution $\hat{x}$ is the intersection of the measurement line $y = Ax$ and the minimum norm-ball in each case. The $l_2$-norm constraint leads to non-sparse solutions, while the $l_p$-norm, $0 < p \leq 1$, constraint promotes sparse solutions.](image)

Nevertheless, $G(x)$ is a concave function and it can be minimized by a majorization-minimization approach rather than a convex minimization as with the $l_1$-norm.

Concisely, minimizing a function $f(x), x \in \Omega$ with the majorization-minimization framework involves finding a convex function $g(x|x_a), x \in \Omega$ which majorizes $f(x)$, such that $g(x|x_a) \geq f(x)$ and $g(x_a|x_a) = f(x_a)$, and then minimizing the function $g(x|x_a)$ with respect to $x \in \Omega$ resulting in $g(x_{a+1}|x_a) = \min_{x \in \Omega} g(x|x_a)$. This procedure assures a descent for $f(x), x \in \Omega$, since $f(x_{a+1}) \leq g(x_{a+1}|x_a) \leq g(x_a|x_a) = f(x_a)$. Repeating this process iteratively until convergence results in minimization of the concave function $f(x), x \in \Omega$.

For a differentiable concave function $f(x), x \in \Omega$ a majorization function can be easily found through the derivative $f'(x) \leq f'(x_0) + f(x_0)(x-x_0)$ and minimized such that $\min_{x \in \Omega} g(x|x_0) = \min_{x \in \Omega} f'(x_0)x$ (the constant terms are ignored as they don’t affect the optimization).

Following the aforementioned analysis, the minimization of a concave function subject to constraints can be recast in an iterative convex minimization problem.

For example, the FOCUSS algorithm [7] minimizes the concave function $G(x) = \sum_{i=1}^{N} |x|^p$ with $0<p<1$. Using the transformation $h_i = x_i^2 \geq 0$ yields $G(h) = \sum_{i=1}^{N} \frac{p}{2} h_i^p$ and $G(h|h_k) = \frac{p}{2} \sum_{i=1}^{N} h_i^{p-1}$. Thus in the k+1 iteration the function to be minimized is
\( G(h|h_k) = \frac{p^N}{2} h_{i,k}^{p-1} h_i \) or equivalently \( \frac{p^N}{2} \left( \hat{x}_i^2 \right)^{p-1} (x_i^2) = \frac{p^N}{2} \|W_k x\|^p_2 \). The matrix 
\[ W_k = \text{diag} \left( \hat{x}_i, k \right) \] 
is diagonal with elements related to the solution of the previous iteration, \( \hat{x}_k \), resulting in the iterative l2-norm convex minimization problem,

\[
\min_{x, C^N} \|W_k x\|^p_2 \text{ subject to } \|Ax - y\|_2 \leq \varepsilon. \quad (P_{2,w}^e)
\]

Reformulating \((P_{2,w}^e)\) in an unconstrained form with the use of Lagrange multipliers,

\[
\min_{x, C^N} \|W_k x\|^p_2 + \eta \|Ax - y\|^2_2, \quad (P_{2,w}^q)
\]

the solution can be found analytically in each iteration,

\[ \hat{x}_{k+1} = W_k^H A^H \left( A W_k^H W_k A^H + \eta I_M \right)^{-1} y. \]

Even though the problem is not convex and convergence to the global minimum is not guaranteed, initialization with the l2-norm solution such that \( W_0 = I_N \) usually suffices (at least in the context of DOA estimation) to assure that \((P_{2,w}^e)\) does not get trapped in local minima.

4. EXPERIMENTAL RESULTS

The high-resolution capabilities and the robustness of the sparsity promoting framework in DOA estimation, is demonstrated on ocean acoustic measurements for source tracking from single snapshot data. Specifically, the data is from the long range acoustic communications (LRAC) experiment collected from a towed horizontal uniform linear array from 10:00-10:30 UTC on 16 September 2010 in the NE Pacific. The array has \( M = 64 \) sensors, with intersensor spacing \( d = 3m \) and was towed at 3.5 knots at 200m depth. The data were acquired with a sampling frequency of 2000Hz and the record is divided in 4s non-overlapping snapshots. Each snapshot is Fourier transformed with \( 2^{13} \) samples.

Figure 2 shows the reconstruction with CBF, CS and l\( p \)-norm minimization, with \( p = 0.8 \), at frequency \( f = 125 \text{Hz} \) \( \left( \frac{d}{\lambda} = \frac{1}{4} \right) \) for a DOA grid \([-90:1:90]^\circ\]. The beamformer output indicates the presence of three sources at around 45\(^\circ\), 30\(^\circ\) and -65\(^\circ\). The two arrivals at 45\(^\circ\) and 30\(^\circ\) are attributed to distant transiting ships. The broad arrival at -65\(^\circ\) is from the towship R/V Melville. The beamforming map suffers from low resolution and artifacts due to sidelobes. The CS provides high resolution imaging, which is further improved by enhancing sparsity with l0.8-norm minimization.
Fig.2: Data from LRAC: (a) conventional beamforming, (b) CS with $l_1$-norm minimization (implemented with CVX [4]), (c) CS with $l_{0.8}$-norm minimization (implemented with FOCUSS [7])

5. CONCLUSION

Sound source localization with sensor arrays is essentially a sparse signal reconstruction problem, which can be efficiently solved with CS or other sparsity promoting optimization procedures providing high-resolution reconstruction in DOA estimation.

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DOA ESTIMATION ALGORITHM AS APPLIED TO WIDEBAND PROCESSING

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Abstract: The paper discusses the algorithm for resolution of signal sources including those with angular separation less than the width of antenna main lobe for the case of wideband signal processing based on the solution of observation equation system. The estimate is shown to be unbiased, and optimal weight coefficients are derived which provide the estimate variance exceeding the minimal possible one (Cramer-Rao bound) by less than 10% on average. The given algorithm form does not require a priori data on the tactical situation.

Keywords: direction of arrival (DoA) estimation, superresolution, Cramer-Rao bound
INTRODUCTION

In passive location, there is sometimes a need to resolve signal sources spaced by the distance less than the width of antenna main lobe. These problems are critical in remote control or deep ocean navigation. Signal processing algorithms providing such resolution are referred to as superresolution algorithms. Their overview is given in [1]. This paper proposes a DoA estimation algorithm not requiring the direction-finding pattern.

The system of observation equations is usually written as:

\[ u_{j,k,t} = \sum_{m=1}^{M} g_{j,k}(\theta_m) a_{j,t,m} + n_{j,k,t} \]  

where \( j \in [1,J] \) is the index of frequency sample, \( k \in [1,K] \) is the number of spatial channel, \( t \in [1,N_t] \) is the index of time sample, \( M \) is the number of signals, \( u_{j,k,t} \) is the complex output of \( k \)-th spatial channel for \( j \)-th frequency sample at \( t \)-th time point, \( a_{j,t,m} \) is the complex amplitude of signal from \( m \)-th source at \( t \)-th time point at \( j \)-th frequency sample, \( \theta_m \) is the DoA of \( m \)-th signal source, \( g_{j,k}(\theta_m) \) is the value of complex directivity pattern of \( k \)-th spatial channel at \( j \)-th frequency sample towards the \( m \)-th signal source, \( n_{j,k,t} \) is the interfering signal in \( k \)-th observation channel for \( j \)-th frequency sample at \( t \)-th time point. Note that we consider only a broadband case, so large volume of time statistical data is not needed; the problem becomes overdetermined after several cycles already.

DoAs of sources \( \theta \) and their amplitudes \( a \) are unknown parameters in the problem; outputs of spatial channels \( u \) are known. The number of equations is \( 2 \cdot J \cdot N_t \cdot K \) (taking into account the complexity of values), and the number of unknown parameters is \( M \cdot (1 + 2 \cdot J \cdot N_t) \). In order to set the estimation problem correctly, the number of unknown parameters should not exceed the number of measurements:

\[ M \cdot (1 + 2 \cdot J \cdot N_t) \leq 2 \cdot J \cdot N_t \cdot K . \]  

PROBLEM STATEMENT AND SOLUTION

Given is the system of equations (1), DoA of sources \( \theta_m\), \( m \in [1,M] \) should be found.

The problem conditions are as follows:

1. The number of sources \( M \) is known;
2. The condition of the system completeness (2) is fulfilled;
3. Complex amplitudes of signals and noise are zero-mean random processes;
4. Complex amplitudes are noncorrelated with each other and with noise.

There exist multiple methods for determining the number of sources, some of them are described in [Error! Reference source not found., Error! Reference source not found.]. The difficulty in real water situation consists in the presence of multipath sound propagation effect, which sometimes results in non-fulfilment of condition 4. Superresolution algorithm for correlated signal sources is given, for example, in [Error! Reference source not found.--Error! Reference source not found.].
The system of equations (1) is linear with respect to signal amplitudes. Assume that there is no noise; then with fixed \( j \) and \( t \), omitting these indices in the notation, the following can be written:

\[
    u_k = \sum_{m=1}^{M} g_{j}(\theta_m) a_m
\]  

Equations (3) are quasilinear, so the regression equation can be written as follows:

\[
    u_{k+M} = \sum_{m=1}^{M} h_{j,m} u_{m+1};
\]  

Here \( k \in [1, K-M] \). Regression coefficients \( h_{j,m} \) will be determined using the observation equations (3) for representation of spatial channel outputs.

\[
    u_{k+M} = \sum_{m=1}^{M} a_m g_{k+M}(\theta_m) = \sum_{m=1}^{M} h_{j,m} \left[ \sum_{m_0=1}^{M} g_{k+1-m_0}(\theta_{m_0}) a_{m_0} \right] = \sum_{m=1}^{M} a_m \left[ \sum_{m_0=1}^{M} h_{j,m_0} g_{k+1-m_0}(\theta_{m_0}) \right];
\]  

Equations (5) will be fulfilled if the following conditions of regression coefficients are satisfied:

\[
    g_{k+M}(\theta_m) = \sum_{m_0=1}^{M} h_{j,m_0} g_{k+1-m_0}(\theta_{m_0});
\]  

From this formula we can directly obtain a vectorial expression for the sought coefficients:

\[
    \vec{h}_j = G^{-1}_j \vec{g}_{j+M},
\]  

where \( \vec{h}_j = (h_{j,1}, \ldots, h_{j,M})^T \), \( \vec{g}_{j+M} = (g_{j,M}(\theta_1), \ldots, g_{j+M}(\theta_M))^T \) are the vectors of length \( M \), \( G_j = \begin{pmatrix} g_j(\theta_1) & \cdots & g_{j+M-1}(\theta_1) \\ \vdots & \ddots & \vdots \\ g_j(\theta_M) & \cdots & g_{j+M-1}(\theta_M) \end{pmatrix} \) is the \( M \times M \) matrix, \( T \) is the transposition symbol. From Eq. (6), DoAs of sources are zeroes of the following function:

\[
    d_k(\theta) = g_{k+M}(\theta) - \sum_{m_0=1}^{M} h_{j,m_0} g_{k+1-m_0}(\theta);
\]  

To make the coordinate estimation more accurate, we shall use the equations for all spatial channels, as well as for previously fixed frequency and time indices, using a linear combination:

\[
    d(\theta) = \sum_{i=1}^{N} \sum_{j=1}^{J} \sum_{k=1}^{K} c_{j,t,k} \left[ g_{j,k+M}(\theta) - \sum_{m_0=1}^{M} h_{j,t,k,m_0} g_{j,k+1-m_0}(\theta) \right]
\]  

Roots of this equation will be the joint estimates of the source DoAs; coefficients \( c_{j,t,k} \) are arbitrary, they should be chosen so that DoA estimation variance is minimal. To solve the stated problem, we should first estimate the regression coefficients \( h_{j,t,k,m_0} \), then generate a function \( d(\theta) \) using the known characteristics of the antenna array, and then find zero of the function, for example, using the well-known Newton method. For
simplicity, divide the problem solution into two steps by presenting function (9) as narrowband and wideband parts:

\[
d(\theta) = \sum_{t=1}^{N_t} \sum_{j=1}^{J} b_{jt} \left[ \sum_{k=1}^{K-M} c_k \left( g_{j,k,M}(\theta) - \sum_{m=0}^{M} h_{j,t,k,m,0} g_{j,k,M-1}(\theta) \right) \right] = \sum_{t=1}^{N_t} \sum_{j=1}^{J} b_{jt} d_{j,t}(\theta)
\]

The sum in brackets in (10) can be rewritten in the vector form, then the function for the search of narrowband estimates takes the following form:

\[
d_{j,t}(\theta) = \sum_{k=1}^{K-M} c_k \left( g_{j,k,M}(\theta) - \tilde{g}_{j,k}^{\mathcal{H}}(\theta) \tilde{h}_{j,k} \right)
\]

where

\[
\tilde{h}_{j,k} = \left( h_{j,t,k,1}, \ldots, h_{j,t,k,M} \right)^T, \quad \tilde{g}_{j,k}(\theta) = \left( g_{j,k}^{\mathcal{H}}(\theta), \ldots, g_{j,k,M-1}^{\mathcal{H}}(\theta) \right)^T, \quad H \text{ is Hermitian conjugation.}
\]

To find linear regression coefficients using outputs of the formed spatial observation channels, we rewrite (4) in matrix form using time samples for collecting the statistics. It is reasonable to use time samples rather than frequency samples because the antenna directivity pattern varies with frequency and not with time; so the solution of the narrowband problem will be greatly simplified. Thereafter in (10), (11) we’ll omit index \( t \).

\[
\bar{u}_{j,k+M} = U_{j,k} \tilde{h}_{j,k},
\]

where

\[
\bar{u}_{j,k+M} = \left( u_{j,k+M,1}, \ldots, u_{j,k+M,N_t} \right)^T, \quad U_{j,k} = \left( \bar{u}_{j,k}, \ldots, \bar{u}_{j,k+M-1} \right),
\]

\[
\tilde{h}_{j,k} = \left( h_{j,k,1}, \ldots, h_{j,k,M} \right)^T.
\]

If \( N_t \geq M \), the vector of regression coefficients can be determined using the least squares method:

\[
\tilde{h}_{j,k} = \left( U_{j,k}^{\mathcal{H}} U_{j,k} \right)^{-1} U_{j,k}^{\mathcal{H}} \bar{u}_{j,k+M}
\]

where \( k \in [1, K - M] \).

If there were no noise, the problem would have been solved already. To take the noise effect into account, consider the limit \( \tilde{h}_{j,k} \) with \( N_t \rightarrow \infty \), with consideration for the fourth condition of the problem:

\[
\tilde{h}_{j,k} \xrightarrow{N_t \rightarrow \infty} \left( G_{j,k}^{\mathcal{H}} W_s G_{j,k} + W_n_j \right)^{-1} G_{j,k}^{\mathcal{H}} W_s G_{j,M+k},
\]

here \( G_{j,k} \) is the matrix of directivity pattern values, \( W_n_j \) is the noise correlation matrix at the \( j \)-th spectral sample, \( W_s_j \) is the signal correlation matrix at the \( j \)-th spectral sample. According to the problem conditions, the signal and noise correlation matrices are diagonal: \( W_s_j = \text{diag} \left( W_{s,j,1}, \ldots, W_{s,j,M} \right) \), \( W_n_j = \text{diag} \left( W_{n0_j}, \ldots, W_{n0_j} \right) \), \( W_{s,j,m} \) is the power of the \( m \)-th signal at the \( j \)-th frequency sample, \( W_{n0_j} \) is the noise power. We assume that noise has the same power in all the channels. Based on the limiting value of vector of regression coefficients (14), we introduce the relevant correction to (13) to compensate for the noise effect.

\[
\overline{hc}_{j,k} = \left( U_{j,k}^{\mathcal{H}} U_{j,k} - W_{n0_j} \cdot I \cdot N_t \right)^{-1} U_{j,k}^{\mathcal{H}} \bar{u}_{j,k+M},
\]

here \( \overline{hc}_{j,k} \) is the correlated vector of regression coefficients. The limiting value of this vector is close to the true value determined by (7):
\[
\overline{h}_{c,j,k} \rightarrow (G_{j,k}^{H} W_{j} G_{j,k})^{-1} G_{j,k}^{H} W_{j} \tilde{g}_{j,k,M+k} = G_{j,k}^{-1} \tilde{g}_{j,k,M+k},
\]

(16)

Thus using the coefficient vector (15) we can generate the function \( d(\theta) \) for some set of coefficients \( c_{k}, b_{j} \), and zeroes of this function will be the solution to the problem.

Let us show that there exists a unique set of coefficients \( c_{k}, b_{j} \), which minimizes the variance of coordinate estimation errors. We introduce several denotations: \( \theta_{e,m} \) is the DoA estimate of \( m \)-th source; \( \Delta \theta_{m} = \theta_{m} - \theta_{e,m} \) is the error of DoA estimate of \( m \)-th source. Vary the function \( d(\theta) \) around the DoA \( \theta_{m} \):

\[
\delta d(\theta)|_{\theta_{0}} = \sum_{j=1}^{J} b_{j} \left[ \sum_{k=1}^{K-M} c_{k} \left( \Delta \theta_{m} \left[ g'_{j,k,M} (\theta_{m}) - \tilde{g}^{HH}_{j,k} (\theta_{m}) \tilde{h}_{j,k} \right] + \tilde{g}^{HH}_{j,k} (\theta_{m}) \Delta \tilde{h}_{j,k} \right) \right] = 0;
\]

(17)

where \( \Delta \tilde{h}_{j,k} \) is the error of regression coefficient vector, \( g'_{j,k,M} (\theta_{m}) = \frac{dg_{j,k,M}(\theta)}{d\theta} \bigg|_{\theta=\theta_{m}} \), \( \tilde{g}^{HH}_{j,k} (\theta_{m}) \) is the error of regression coefficient vector. Represent (17) in the other form:

\[
\delta d(\theta)|_{\theta_{0}} = -\sum_{j=1}^{J} b_{j} \left[ \sum_{k=1}^{K-M} c_{k} \left( g'_{j,k,M} (\theta_{m}) - \tilde{g}^{HH}_{j,k} (\theta_{m}) \tilde{h}_{j,k} \right) \right] \Delta \theta_{m} + \left( \delta d_{j}(\theta)|_{\theta_{0}} \right)_{h} \Delta h_{j,k} = 0,
\]

(18)

where \( \delta d_{j}(\theta)|_{\theta_{0}} = \sum_{k=1}^{K-M} c_{k} \left[ g'_{j,k,M} (\theta_{m}) - \tilde{g}^{HH}_{j,k} (\theta_{m}) \tilde{h}_{j,k} \right] \), \( \left( \delta d_{j}(\theta)|_{\theta_{0}} \right)_{h} \Delta h_{j,k} = \sum_{k=1}^{K-M} c_{k} \tilde{g}^{HH}_{j,k} (\theta_{m}) \Delta \tilde{h}_{j,k} \)

Solution to the problem (18), which minimizes DoA variance in narrowband case with \( J=1 \) is known \[5\] and is written as follows:

\[
\left( \Delta \theta_{m} \right)^{2} = D_{j} = \frac{\left( \delta d_{j}(\theta)|_{\theta_{0}} \right)_{h}^{k} \Delta h_{j,k}}{\left( \delta d_{j}(\theta)|_{\theta_{0}} \right)_{h}^{2}} = \frac{1}{(Nt-M)} \frac{1}{\bar{p}^{H} Y^{-1} \bar{p}}.
\]

(19)

The following denotations are used:

\[
Y_{k,k'} = \frac{W_{n} \theta_{j,k'}}{W_{n} \theta_{j,k}} \left(1+\left| \tilde{h}_{j,k} \right|^{2} \right);
\]

\[
Y_{k,k'} = 0;
\]

\[
(\bar{p})_{k} = g'_{j,k,M} (\theta_{m}) - \tilde{g}^{HH}_{j,k} (\theta_{m}) \tilde{h}_{j,k};
\]

Express \( \Delta \theta_{m} \) from (18), take a square of its module and average over noise and random amplitudes to get the estimate of variance

\[
\left( \Delta \theta_{m} \right)^{2} = \frac{\sum_{j=1}^{J} b_{j} \left( \delta d_{j}(\theta)|_{\theta_{0}} \right)_{h}^{k} \Delta h_{j,k}}{\left( \sum_{j=1}^{J} b_{j} \delta d_{j}(\theta)|_{\theta_{0}} \right)_{h}^{2}}
\]

(20)
Estimating $\Delta h_{j,k}$ in the first approximation and minimizing (20) by coefficients $b_j$, we find the optimal set of coefficients. With the mathematical derivations omitted, the set is given by

$$b_0 = \left( \begin{array}{c} 1/d_1(\theta)^H \cdot D_i \\ \vdots \\ 1/d_j(\theta)^H \cdot D_j \end{array} \right) \quad (21)$$

Using optimal values of coefficients $c_k$ [5] and omitting the constant multiplier, it can be easily found that all coefficients are equal to relevant signal to noise ratios at the specified frequencies:

$$b_0 = \left( \frac{W_{S_{j,m}}}{W_{n_0}}, \ldots, \frac{W_{S_{j,m}}}{W_{n_0}} \right) \quad (22)$$

Substituting (22) to (20) yields the optimal variance of DoA estimate for the $m$-th source

$$\overline{(\Delta \theta^2_m)} = \frac{1}{Nt - M} \left( \sum_{j=1}^{J} \frac{W_{S_{j,m}}}{W_{n_0}} \sum_{k=1}^{K-M} \left[ g'_{j,k+M}(\theta_m) - \bar{g}^{H\prime}_{j,k}(\theta_m) \bar{h}_{j,k} \right]^2 \left( 1 + \|\bar{h}_{j,k}\|^2 \right)^{-1} \right)^{-1} \quad ; \quad (23)$$

Expression (10) for DoA of sources with optimal sets of coefficients is written as

$$d(\theta) = \sum_{j=1}^{J} \frac{W_{S_{j,m}}}{W_{n_0}} \sum_{k=1}^{K-M} \left( g'_{j,k+M}(\theta_m) - \bar{g}^{H\prime}_{j,k}(\theta_m) \bar{h}_{j,k} \right)^H \left( g_{j,k+M}(\theta) - \bar{g}^{H\prime}_{j,k}(\theta_m) \bar{h}_{j,k} \right); \quad (24)$$

Obviously, in (24) DoA of the $m$-th source $\theta_m$ can be substituted with variable $\theta$, because DoA estimate remains unbiased and its variance does not change. Standard deviations of estimates for narrowband and wideband cases are compared in Fig. 1.

In real conditions, we do not have deep knowledge of the environment and tactical situation, so some approximations are used:

- signal power is taken to be constant $W_{S_{j,m}} = W_{S_0}$.
Fig. 1. Standard deviation normalized to the limit values determined by the Cramer-Rao bound vs angular separation between two sources, normalized to the width of antenna main lobe
a) narrow band; b) broad band;

- noise power is taken to reduce by 6 dB at each octave: $W_{n0,j} = W_{n0} \frac{f_j^2}{f_j^2}$, where $f_j$ is the frequency corresponding to the $j$-th frequency sample.

The formula that can be applied in practice is given below:

$$d(\theta) = \sum_{j=1}^{J} \sum_{k=1}^{K-M} \left( g_{j,k+M}^T(\theta) - \tilde{g}_{j,k}^M(\theta) \tilde{h}_{j,k} \right)^H \left( 1 + \| \tilde{h}_{j,k} \|^2 \right) \cdot \left( g_{j,k+M}(\theta) - \tilde{g}_{j,k}^M(\theta) \tilde{h}_{j,k} \right).$$  \hspace{1cm} (25)

To conclude the paper we should note that the developed algorithm works with all types of antenna arrays. However, in case of linear equidistant antennas the problem is greatly simplified because (25) presents a polynomial of power $M$, whose zeroes can be easily found. Nevertheless, preliminary focusing procedure, which is rather labour-consuming, is required.

References
MULTIVARIATE DISTRIBUTIONS OF CLUTTER LEVELS FOR AUTOMATED CLASSIFIERS

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Abstract: Active sonar automated classifiers begin by identifying potential targets in a set of acoustic data. These candidates must pass a detection test of having sufficient level above the detection threshold. The levels at these points also must be a local maximum. Once candidate target locations have been selected a small snippet of data is excised centered about each candidate point. The behavior of the time series within these snippets is used to classify signals of interest from clutter. The distribution of data levels within the snippets may be required to define an optimal classifier. A general formula for the conditional distribution of levels is derived given that a point is within the snippet. The conditional distributions are a function of the distance of the point from the central peak, and depend on the bivariate distribution of pairs of points prior to snippet selection. We compare these analytic distributions with data taken from the NATO Base 04 exercise. Before snippets are formed, the prior data intensities are well modeled with generalized Pareto distribution marginals and a bivariate Gumbel copula.

Keywords: active sonar, classifier, distribution, copula, clutter, generalized Pareto distribution
BACKGROUND

Active sonar automated classifiers begin by identifying potential targets in a set of acoustic data. These candidates must pass a detection test of having sufficient level above the detection threshold. The levels at these points also must be a local maximum. Once a candidate location has been selected, a small snippet of data is excised centered about the candidate target point. The behavior of the time series within these snippets is used to classify signals of interest from clutter. One algorithm of interest is the adaptive coherence estimator (ACE) [1]. This has been successfully demonstrated as a featureless classifier on active sonar data [2]. However, the ACE derivation assumes that the clutter data are Gaussian distributed. It is well known that active sonar data clutter points are best characterized as non-Gaussian with much heavier upper tails [3-5]. In addition, the process of selecting snippets biases the data distributions, when the attention is restricted to snippets alone. Clearly, the distribution of levels at the center point of the snippets cannot have any values below the detection threshold, making it quite different from the distribution across all data before snippets are identified. Furthermore, points near the snippet center tend to be correlated to the center point, which also biases these at higher levels than the distribution across all data. A model for the distributions of the snippet points is required in order to modify the ACE algorithm or derive an alternative to account for the non-Gaussian nature of the clutter distribution. This could potentially improve the performance of the automated classifier. A general model is derived here to account for the biasing effects of selecting the snippet data.

When several random variables are all jointly Gaussian, their joint distribution function is the well-known multivariate Gaussian. This distribution is completely described by two moments: the mean of each variable and the covariance matrix. When several random variables are non-Gaussian the situation can be much more complex. Two moments may not provide a complete description. Furthermore, in general, a single analytic function may not be available or known to completely describe all the mutual joint interactions. One technique to simplify the description of multivariate distributions is the copula [6]. Each random variable is transformed to a uniformly distributed random variable on the interval [0, 1]. By relating the uniformed random variables to one another, the copula represents the joint distribution of these uniform random variables and contains all the information required to describe the joint distribution of the original non-Gaussian random variables.

DISTRIBUTION MODEL DERIVATION

Let X and Y represent the acoustic intensity at two points within a time series at some separation, before any snippets are made. Let X and Y also have some prior known joint distribution with cumulative distribution \( F_{XY}(x,y) \) and density \( f_{XY}(x,y) \). Furthermore, assume stationarity, so that the marginal of X is the same as the marginal of Y with distribution \( F_Y(y) \).

In the process of making a snippet, we only use data centered at points above some threshold value, \( x_0 \). Now let X be the random variable representing level at the center point of the snippet. We want to know what the conditional distribution of y is at some separation in time from the peak, after the snippets are made. In constructing the snippet
we also require that $X$ be a local maximum, so we will assume that $Y < X$, for all points within the snippet.

Let $F_Y|X(y|x)$ be the cumulative conditional distribution of $Y$ given that $X$ is above the threshold, and $Y < X$. From the conditional probability relation

$$F_{Y|X}(y|x) = \Pr(Y < y \mid x_0 < x \& Y < x) = \frac{\iint_{R} f_{XY} \, dx \, dy}{\iint_{R_{\infty}} f_{XY} \, dx \, dy}$$

The region $R$ is defined by the area for which $x_0 < x$, $Y < y$, and $Y < x$, shown in Fig. 1, and the region $R_{\infty}$ is the region $R$ when $y = \infty$.

![Fig. 1. Areas of integration](image)

Evaluating the integrals over these regions yields

$$F_{Y|X}(y|x) = \begin{cases} 
    \frac{F_Y(y) - F_{XY}(x_0, y)}{1 - F_{XY}(x_0, x_0) - \int_{x_0}^{\infty} \frac{\partial}{\partial y} F_{XY}(x, y) \bigg|_{x=y} \, dy}, & y \leq x_0 \\
    \frac{F_Y(y) - F_{XY}(x_0, y)}{1 - F_{XY}(x_0, x_0) - \int_{x_0}^{y} \frac{\partial}{\partial y} F_{XY}(x, y) \bigg|_{x=y} \, dy}, & y > x_0 
\end{cases}$$

To proceed further analytically, we need an explicit formula for $F_{XY}(x, y)$. One choice is the Gumbel copula case [6] with parameter $\theta$:

$$F_{XY}(x, y) = \exp \left\{ - \left[ (-\ln(F(x)))^\theta + (-\ln(F(y)))^\theta \right]^\frac{1}{\theta} \right\}$$

In the following section this choice will been shown to match well with the bivariate distribution of data points prior to snippet formation. Here $F$ is the prior marginal cumulative density of the original data (since we assume prior stationarity, $F_Y(y)$ and $F_X(x)$ are the same distribution). Evaluation of the derivatives and integrals can now be performed yielding the conditional distribution for points within the snippet:
Although the above formulas could be used with any marginal densities, the generalized Pareto distribution (GPD) has often been found to represent the marginal distribution of real data well [3, 5]:

\[
F_{Y|X}(y|x) = \begin{cases} 
\frac{F(y) - F_{XY}(x_0, y)}{\frac{1}{2}[1 - F^{2^{1/\theta}}(x_0)]}, & y \leq x_0 \\
\frac{F(y) - \frac{1}{2}[F^{2^{1/\theta}}(y) + F^{2^{1/\theta}}(x_0)]}{\frac{1}{2}[1 - F^{2^{1/\theta}}(x_0)]}, & y > x_0 
\end{cases}
\]

This gives an analytic expression with three parameters for how the distribution of snippet intensity changes at any point as the time separation from the center point increases. The two GPD parameters are determined by fitting the prior distribution of acoustic intensity, before the snippets are formed. As shown in the following section, the Gumbel parameter \( \theta \) is the key parameter that will change with separation. Presumably how \( \theta \) changes may also depend on sample rate, bandwidth, and perhaps other environmental factors for each data set, and will have to be determined. In general \( \theta \) is very large near the center point and approaches 1 as separation increases and subsequently the data become uncorrelated to the central point.

**DATA COMPARISON**

Comparisons are made here with data from the NATO Base 04 exercise, conducted on the Malta Plateau [7]. One ping on Julian day 152, 2004 at 14:24 was processed. A 60 Hz wide subband from 1240 to 1300 Hz of the transmitted FM signals was matched filtered and normalized to remove large scale variations. A section of data dominated by bottom clutter from the Ragusa Ridge was isolated. The clutter region was defined by time from 7 to 17 s after transmission, and beams 12 to 65 out of 120 beams spanning 360 deg.

To evaluate the validity of our analytical model, we compared an empirical data copula fit to a copula developed using the analytic model with GPD marginals. The fitting process finds the Gumbel parameter that minimizes the sum of the squared differences between the empirical and model copulas over all their points. This produces an excellent fit for our data as shown in Figs. 2 and 3. When the data are well separated, they are nearly uncorrelated and the Gumbel parameter is near to 1, as in Fig. 2. When the data are nearby, as in Fig. 3, the Gumbel parameter is higher. This agreement justifies our use of the Gumbel form in the derivation of the previous section.
Using the least square fitting method, the Gumbel parameter variation with data separation time, $\Delta t$ was determined. This is a non-Gaussian analog of the correlation function for the Gaussian case. For these data we observe that a good approximation is a power law, shown in Fig. 4:

$$\theta - 1 = a \Delta t^b$$

The best fit power law parameters are $a = 0.0015$ and $b = -1.4$ for these data. The joint distribution is defined by the copula and the marginal distribution. The marginal is well fit by a generalized Pareto distribution (GPD). Note that the GPD model is applied to acoustic intensities, and was transformed to the corresponding dB levels for the comparison in Fig. 5. A primary motivation for the GPD model is the power law behavior of the upper tail. The upper tail is better observed by plotting the complementary cumulative density (one minus the cumulative density or false alarm rate) on a log plot as in Fig. 5. Both the data and the GPD model approach a straight line asymptote for high clutter levels in this plot.
With the prior joint distribution completely characterized by the marginal distribution and the Gumbel copula, the theory for the distribution after snippets have been taken can be compared to the data. To form the snippets, 201 data points (0.2 s) were centered on each candidate peak above a threshold of 6 dB. A total of 735 clutter snippets were found over the Ragusa Ridge region for this ping. Fig. 6 shows 3 representative cases for different separations from the peak. When the data are well separated from the peak, the distribution is nearly the same as the prior, although the upper tail is reduced since it cannot surpass the peak of the snippet. This agrees well with the model (black curves in...
Fig. 6. When the data are near the peak, the distribution is sharply peaked with nearly no lower tail below the threshold value (blue curves in Fig. 6). Again, there is good agreement with the model distribution. However, the agreement with the model is not good for intermediate values such as the red curves in Fig. 6. The data are more sharply peaked with a shorter lower tail. We must conclude that the prior distribution we used across all data is not representative of data within the snippets. In particular, the higher level clutter objects captured in the snippets may have more structure and higher correlation than the general clutter data. Therefore, the joint density distributions must be reexamined for data confined to the snippets.

The bivariate copulas were examined for data within the snippet as a function of separation from the peak. These copulas were still well matched to the Gumbel copula model, however, the Gumbel parameter had a different behavior than it did in characterizing all the data prior to snippet formation. The power law model is still appropriate but the parameters now have the new values $a=0.00026$ and $b=-2$. Fig. 7 shows the improved comparison between the model and data distributions, when the model is based on the Gumbel fit only to data within the snippets.

**SUMMARY**

The distribution of clutter data was derived given that the data are selected to be within a snippet used in automated classification. A model was developed based on the Gumbel copula and the prior distribution of clutter levels before snippets are formed. The model compares well to data, but only after the Gumbel parameter was chosen to represent only the data within snippets, which differs from the parameter when distributions are taken over all data.
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REFERENCES


SEAFLOOR CLASSIFICATION USING STATISTICAL MODELING OF WAVELET SUBBANDS

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Abstract: This paper deals with the classification of textured seafloor images recorded by sidescan sonar. To address this problem, a supervised classification approach based on the Bayesian framework is proposed. In this way, the textured images are characterized through parametric probabilistic models of the wavelet coefficients. The generalized Gaussian distribution (GGD), which is a well-established model to characterize the marginal distributions of the wavelet subbands, is considered. However, to take into account the joint statistics of wavelet coefficients, we also consider the Gaussian copula based multivariate generalized Gaussian model (GC-MGG). A supervised learning context is adopted for the classification stage by using a probabilistic k-Nearest Neighbors classifier. Each textured image will be represented by its GGD or GC-MGG estimated parameters and given a collection of training images the Kullback-Leibler divergence is used to estimate the similarity between a test image and seafloor classes. Experiments on real sonar textured images are proposed to highlight the interest of this approach.

Keywords: Seafloor classification, texture analysis, wavelet transform, statistical modelling.
1. INTRODUCTION

This paper deals with the classification of textured images recorded by sidescan sonar with respect to the sea bottom types. Various approaches of this topic has been proposed and are all relying on the extraction of features from small textured images followed by a supervised or unsupervised classification [1]. Classical methods for extracting the textured seafloor information are based on first order statistics [2], on second order statistics like Grey Level Co-occurrence Matrix [3], [4] and methods based on spectral analysis via filter banks or transforms (Gabor filters, wavelet, Fourier transform...) [5].

In this study, to characterize the sonar textured images, two parametric probabilistic models of the wavelet coefficients are studied for their ability to fit the observed data. The first one is the generalized Gaussian distribution (GGD), which is a well-established univariate model to characterize the marginal distributions of the wavelet subbands coefficients. The second is the Gaussian copula based multivariate generalized Gaussian (GC-MGG) model which takes into account the joint statistics of the wavelet subbands coefficients.

A supervised classification approach based on the Bayesian framework is proposed in this paper. Each textured image being represented by its GGD or GC-MGG estimated parameters, the Kullback-Leibler divergence (KLD) is used to estimate the similarity between a test image and the learned seafloor classes.

In what follows, section 2 gives an overview about classification in the Bayesian framework. The probabilistic models of the wavelet subbands coefficients are described in section 3. The expressions of the KLD are given in section 4. Section 5 shows experimental results obtained on a sonar texture database. Finally section 6 gives some concluding remarks.

2. CLASSIFICATION WITHIN A BAYESIAN FRAMEWORK

Let us consider $N$ classes $C_1, C_2, ..., C_N$ and $\mathbf{w} = (w_1, ..., w_L)$ an observation of $L$ features to be classified. In our study $\mathbf{w}$ denotes samples from a wavelet subband. By associating a class indicator variable $Y \in \{1, ..., N\}$ to classes’ indexes, the optimal decision rule that minimizes the probability of misclassification is given by the well-known maximum a posteriori probability (MAP) decision rule:

$$\text{Classe}(\mathbf{w}) = \arg \max_{i=1,\ldots,N} P(Y = i | \mathbf{w})$$

(1)

It can be shown that this decision rule is asymptotically equivalent to minimizing the Kullback-Leibler divergence, and using a parametric approach we obtain:

$$\text{Classe}(\mathbf{w}) = \arg \min_{i=1,\ldots,N} KLD(p(\mathbf{w}; \theta_w) || p(\mathbf{w}; \theta_i))$$

(2)

where $p(\mathbf{w}; \theta_w)$ and $p(\mathbf{w}; \theta_i)$ are the probability density functions associated with the observation $\mathbf{w}$ to be classified and the class $C_i$ respectively; $\theta_i$ and $\theta_w$ stand for the parameters of the estimated model.

The issue of the Bayesian classification is then to define the probability functions that model the data and to derive the analytical expressions of the KLD for the similarity measurement.
3. MODELLING OF THE WAVELET SUBBAND COEFFICIENTS

Our approach consists in modeling the data in the wavelet domain. In this paper we propose to use two different models.

3.1. Generalized Gaussian distribution modelling

The generalized Gaussian distribution (GGD) is widely used to model wavelet coefficients because of the almost heavy tailed aspect of the empirical probability density functions (PDFs) of subbands [6]. Since the marginal statistics of wavelet coefficients are highly non-Gaussian, the GGD is adequate to represent either the leptokurtic or platykurtic behavior of the marginal distributions of subbands. It is defined by

\[
    f_X(x) = \frac{\beta}{2\alpha \Gamma(1/\beta)} e^{-\left(\frac{\|x\|}{\alpha}\right)^\beta}, \quad x \in \mathbb{R} \quad (3)
\]

where \(\alpha > 0\) and \(\beta > 0\) are the parameters of the distribution and \(\Gamma\) is the Gamma function. \(\alpha\) stands for the width of the PDF peak (also called the scale parameter), while \(\beta\) is inversely proportional to the decreasing rate of the peak (also called the shape parameter). These two parameters are estimated from the maximum likelihood estimation [7] using the Newton-Raphson method [6].

3.2. Gaussian copula multivariate modelling

The Gaussian copula based multivariate generalized Gaussian (GC-MGG) model is a multivariate extension of the GGD [8]. The interest is to take into account the wavelet coefficients dependency. Three types of dependencies can be considered to model the joint statistics of the wavelet coefficients: intra-band, inter-band and inter-scale. It has been shown that in general the predominant part of the dependency corresponds to the spatial one, i.e. the intra-band neighborhoods structure.

After image decomposition into subbands at multiple scales and orientations, neighbors clustered around a reference coefficient are gathered into a \(d\)-dimensional column vector \(X = (x_1, ..., x_d)^t\) where \(d\) is the neighborhood size. Under the spatial homogeneity assumption of each subband, multiple observations of vector \(X\) are obtained by moving a window across subbands and the samples of the \(d\)-dimensional vector are noted \(X_k\), with \(k = 1, ..., L\), where \(L\) is the size of a wavelet subbands. The joint PDF is defined as

\[
    f_X(x_1, ..., x_d) = c(F_1(x_1), ..., F_d(x_d)) \prod_{i=1}^{d} f_i(x_i) \quad (4)
\]

where \(c : [0,1]^d \rightarrow [0,1]\) is the density of the copula, \(f_i\) and \(F_i, i = 1, ..., d\) are respectively the marginal PDFs and the cumulative distributions.

The Gaussian Copula is of practical choice. The expression of the GC-MGG is obtained by using in the joint PDF the marginals and the cumulative distribution function (CDF) of the GGD.
The density of the copula is given by:

\[ c(u_1, \ldots, u_d) = \frac{1}{|\Sigma|^{1/2}} e^{-\frac{\gamma^T (\Sigma^{-1} - I_d) \gamma}{2}} \tag{5} \]

where \( I_d \) is the identity matrix, \( \Sigma \) is the covariance matrix, \( y = (y_1, \ldots, y_d)^T \) is a vector of normal scores such that \( y_i = \phi^{-1}(u_i), i = 1, \ldots, d \), and \( \phi \) is the CDF of the normalized Gaussian distribution \( N(0,1) \).

4. **KULLBACK-LEIBLER DIVERGENCE**

With respect to the decision rule given by equation (2), we use a probabilistic k-Nearest Neighbors classifier. Hence we need to define the associated KLD for both models. However, the KLD is not symmetric that is why we use the Jeffreys Divergence (JD), which is the symmetrized version of the KLD, as similarity measurement.

The JD between two GGD distributions \( f_X(x; \alpha_1, \beta_1) \) and \( g_X(x; \alpha_2, \beta_2) \) is [6]:

\[ JD(f_X\|g_X) = \left( \frac{\alpha_1}{\alpha_2} \right)^{\frac{1}{\beta_2}} \frac{\beta_2}{\beta_1} \Gamma \left( 1 + \frac{\beta_1}{\beta_2} \right) + \left( \frac{\alpha_2}{\alpha_1} \right)^{\frac{1}{\beta_1}} \frac{\beta_1}{\beta_2} \Gamma \left( 1 + \frac{\beta_2}{\beta_1} \right) - \frac{1}{\beta_1} - \frac{1}{\beta_2} \tag{6} \]

The JD between two GC-MGG distributions \( f_X(x; \alpha_{i,1}, \beta_{i,1}, \Sigma_1) \) and \( g_X(x; \alpha_{i,2}, \beta_{i,2}, \Sigma_2) \) is [8]:

\[ JD(f_X\|g_X) = \sum_{i=1}^{d} \left( \frac{\alpha_{i,1}}{\alpha_{i,2}} \right)^{\frac{1}{\beta_{i,2}}} \frac{\beta_{i,2}}{\beta_{i,1}} \Gamma \left( 1 + \frac{\beta_{i,1}}{\beta_{i,2}} \right) + \left( \frac{\alpha_{i,2}}{\alpha_{i,1}} \right)^{\frac{1}{\beta_{i,1}}} \frac{\beta_{i,1}}{\beta_{i,2}} \Gamma \left( 1 + \frac{\beta_{i,2}}{\beta_{i,1}} \right) - \frac{1}{\beta_{i,1}} - \frac{1}{\beta_{i,2}} \tag{7} \]

+ \( 0.5 \left( tr(\Sigma_2^{-1} \Sigma_1) + tr(\Sigma_1^{-1} \Sigma_2) \right) - d \)

5. **EXPERIMENTATION**

5.1. **Sonar texture database**

A database of small images (200x200 pixels) has been compiled from Klein 5500 data recorded during the Battlespace Preparation campaign carried out by the SACLANT Undersea Research Center in La Spezia, Italy. This database is composed of six classes: silt, sand, Posidonia, vertical ripples, 45° ripples, and rock. Some samples are presented in Fig. 1. Each class gathers 60 images characterized by various incident angles and scaling.

5.2. **Experimental results**

In our experiments, the database is divided into two parts: the learning set and the test set. Four different set dimensions are considered and the images in the learning set are chosen randomly. The classification results given in Table 1 are averages of 100 random
combinations. We also provide the results using a classical Bayesian classifier based on the GGD descriptors. The best results are obtained with the GC-MGG model.

**Fig.1: Examples of sonar texture images from the database.**

<table>
<thead>
<tr>
<th></th>
<th>1/8</th>
<th>1/4</th>
<th>1/3</th>
<th>1/2</th>
</tr>
</thead>
<tbody>
<tr>
<td>GGD</td>
<td>90.0</td>
<td>93.1</td>
<td>94.0</td>
<td>95.1</td>
</tr>
<tr>
<td>GC-MGG</td>
<td>89.9</td>
<td>93.7</td>
<td>94.2</td>
<td>95.3</td>
</tr>
<tr>
<td>Naive Bayes</td>
<td>89.0</td>
<td>92.5</td>
<td>92.9</td>
<td>93.6</td>
</tr>
</tbody>
</table>

*Table 1: Classification results for three classification schemes.*

Table 2 provides the confusion matrix for the GC-MGG model using 1/3 of the data as learning set. We can note that the class ‘Rock’ is well classified but the classes ‘sand’ and ‘silt’ show, as expected (since they are visually very close), confusing classifications.

<table>
<thead>
<tr>
<th></th>
<th>Posidonia</th>
<th>45° ripples</th>
<th>Vertical ripples</th>
<th>Rock</th>
<th>Sand</th>
<th>Silt</th>
</tr>
</thead>
<tbody>
<tr>
<td>Posidonia</td>
<td>98.2</td>
<td>0</td>
<td>0</td>
<td>0</td>
<td>0</td>
<td>1.8</td>
</tr>
<tr>
<td>45° ripples</td>
<td>5.5</td>
<td>91.4</td>
<td>2.4</td>
<td>0</td>
<td>0</td>
<td>0.7</td>
</tr>
<tr>
<td>Vertical ripples</td>
<td>0</td>
<td>0.6</td>
<td>97.8</td>
<td>0</td>
<td>1.5</td>
<td>0.1</td>
</tr>
<tr>
<td>Rock</td>
<td>0</td>
<td>0</td>
<td>0</td>
<td>100</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td>Sand</td>
<td>0</td>
<td>0</td>
<td>0</td>
<td>0</td>
<td>88.2</td>
<td>11.8</td>
</tr>
<tr>
<td>Silt</td>
<td>0.3</td>
<td>0</td>
<td>0</td>
<td>0</td>
<td>9.8</td>
<td>89.9</td>
</tr>
</tbody>
</table>

*Table 2: Confusion matrix of the GC-MGG based classifier.*

Several experiments, which cannot be related in this short paper, have been done in order to analyze the behavior of the proposed method and the obtained results. They show that the ripples classes do not always contain (in all images) exact vertical or 45° ripples which leads to wrong classification. The sand and the silt classes are visually closed textures. It appears that both textures are similar (using our models) and the main differences come from the image intensity level. We expect to enhance our classification results by taking into account the marginal distribution of the approximation coefficients which appear to confer discrimination.

Note that we have studied and validated the goodness-of-fit using the classical Kolmogorov-Smirnov test for the GGD and an appropriate multivariate test for the GC-MGG model.
6. CONCLUSION

In this paper, a Bayesian framework has been used to classify texture images grabbed from sidescan sonar images. The classification performances depend on the accuracy of the data models. We have used in this paper the GGD and the GC-MGG models to characterize the statistics of the wavelet subband coefficients. Then, we have derived the expressions of the corresponding Kullback-Leibler divergences used as similarity measurement. The obtained classifications results show the interest of the proposed approach for sonar seafloor classification.

7. ACKNOWLEDGEMENTS

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REFERENCES

Abstract: Classification and tracking are two important techniques for enhancing active sonar performance. Classification rejects unwanted clutter using echo analysis, and tracking provides a history of target motion while rejecting clutter that doesn’t support realistic target motion. Continuous active sonar (CAS) has been proposed as an alternative to conventional pulsed active sonar (PAS), largely in order to provide tracking updates at a much higher rate than is possible with PAS. Unfortunately, these faster updates come at the cost of reduced classification performance, at least for CAS that uses linear frequency modulated waveforms. In this case, maximizing the update rate requires sub-band processing. Classification of echoes from these sub-bands is expected to be relatively poor, since the full bandwidth is favoured for classification. An alternate processing scheme for CAS uses full-band processing, which is typically used for PAS. This potentially maximizes classification performance rather than providing faster updates as in the sub-band approach. A risk of this scheme is the potential for complications in echo signals arising from coherence loss caused by the long duration of CAS waveforms. One facet of a recent Canada-U.S. sea trial, TREX13, focused on conducting experiments that allow direct comparison of the performance of CAS and PAS in shallow water. In this paper, DRDC’s echo classification software was tested with sonar echoes from TREX13. The software, which was originally developed for PAS applications, was used to evaluate whether CAS echoes can be classified as accurately as PAS echoes.

Keywords: active sonar, continuous active sonar, detection, classification
1. INTRODUCTION

Active sonar is required to detect underwater targets that are either silent or too quiet to reliably detect using passive sonar. Active systems, however, can be less effective in shallow littoral environments due to increased false alarms caused by echoes from the seabed in these areas. Two key methods of reducing clutter to improve target detection are classification and tracking. Classification rejects unwanted clutter using signal analysis, while tracking rejects clutter by removing localized contacts that don’t support realistic target motion.

Continuous active sonar (CAS) is an alternative to commonly used pulsed active sonar (PAS). One of the potential advantages of CAS is that it can provide tracking updates at a much higher rate than is possible with PAS. PAS can achieve target detection at most once per ping repetition interval (PRI), which is typically on the order of tens of seconds in order to provide a useful search radius. CAS, on the other hand, can provide many detections within one PRI, potentially obtaining target detections at rates faster than once per second. Unfortunately, these faster updates come at the cost of reduced classification performance and/or reduced signal-to-noise ratio (SNR). In the case of swept waveforms, maximizing the update rate requires sub-band processing, or segmenting the CAS pulse and processing each segment as an individual matched filter. Classification of echoes from these sub-bands is expected to be relatively poor because higher bandwidth is favoured for classification. This dependence on bandwidth was confirmed to be the case for DRDC’s aural classifier [1], which will be used for classifying sonar echoes in this paper.

An alternate processing scheme for CAS applies a matched filter with the full-band replica as is commonly used in PAS. This is the approach used in this paper. This potentially maximizes classification performance rather than providing faster updates as in the sub-band approach. A risk of this scheme is the potential for complications in echo signals arising from coherence loss caused by the long duration of CAS waveforms. In the simplest case, coherence loss would result in lower SNR, which tends to lower classification performance [2]. Of greater concern, however, is the potential for complicated changes on signal features. These changes are difficult to predict and could result in reduced classification capabilities for CAS.

In May 2013, the ONR-sponsored TREX13 sea trial was conducted off the coast of Panama City, Florida. DRDC’s focus during the trial was conducting experiments that allow direct comparison of the performance of CAS and PAS in shallow water. This paper reviews two TREX13 experiments, and compares classification results of full-band PAS and CAS echoes from the experiments using DRDC’s automatic aural classifier, which has not previously been tested on CAS data.

2. EXPERIMENTS AT SEA

2.1. Setup

The TREX13 sea trial was held within 8 km of shore in Panama City, Florida. University of Delaware’s Research Vessel RV SHARP was positioned in a four-point mooring less than 3 km from shore at approximately 30.0599° N, 85.6811° W. The water depth was quite shallow at approximately 20 m over the trial area. The Five Octave
Research Array (FORA) of Pennsylvania State University’s Applied Research Laboratory was positioned 70 m south of SHARP and oriented with a heading of 358°. An ITC 2015 source was positioned 20 m south of SHARP. It was verified that the CAS pulses transmitted by the ITC 2015 did not saturate FORA in this configuration. This is a requirement for CAS operation because pulse transmission and echo reception must occur simultaneously.

The two experiments considered in this paper took place on May 10th, 2013. The runs were each one-hour long with one hour in between. During the PAS run (trex13-r82), a 0.5 s LFM sweep from 1800–2700 Hz was transmitted with a 20 s PRI. A CAS run (trex13-r80) was also performed, where an 18 s LFM swept over the same band was transmitted with the same 20 s PRI. The 2 s down time in the cycle was required for processing time in an echo repeater system used during the trial, resulting in a nearly continuous 90% duty cycle.

DRDC’s CFAV QUEST also participated in TREX13. During each run, QUEST travelled along either the ‘clutter’ or ‘reverb’ track, as depicted in Fig. 1. For the runs considered in this paper, QUEST started near SHARP, and opened at a constant speed of 5 kn at heading 240° along the clutter track. QUEST successfully operated an echo repeater that could repeat continuous transmissions with very low latency and impart a target impulse response on incident signals to simulate target echoes. Four echo repeater techniques were proposed in [3], and DRDC technical staff completed the hardware implementation of all four techniques for the trial. Some echo repeater signals are shown in the next section; however, early on during the trial it became apparent that the hull of QUEST also offered a strong echo, providing a target of opportunity. Furthermore, since this was the first attempt to compare CAS and PAS using the echo repeater, a simple ideal reflector was used for the echo repeater’s impulse response for most of the trial rather than a target response. Therefore, the authors chose to focus on the echoes from QUEST’s hull for the classification results presented in this paper. The data processing used to extract the QUEST echoes is presented next.

![Fig.1: Setup of TREX13 trial area off Panama City, Florida.](image-url)
2.2. Data Processing

After beamforming, the PAS and CAS data were each matched filtered using their respective full-band replicas. Note that CAS processing would normally employ some form of sub-band processing in order to increase the potential number of target detections per ping. This would reduce risk of coherence loss by reducing the processing bandwidth and time; however, the lower bandwidth and lower expected SNR of sub-band echoes would be expected to reduce classification performance based on previous results with DRDC’s aural classifier [1,2]. Therefore, the full-band replicas were used for matched filtering.

An automatic detector was employed after matched filtering. ‘Clutter-mitigation’ images (see Fig. 2) were then formed by plotting each ping as a vertical column of pixels whose brightness was proportional to the enveloped, matched-filter outputs. A sequence of pings formed the horizontal extent of each image. Only the beam corresponding to QUEST’s bearing from FORA is displayed for each ping. The clutter-mitigation images for the PAS and CAS runs are shown in Fig. 2(a) and (b), respectively. A clear trace of QUEST linearly increasing range from FORA can be observed. The echo repeater signal can be seen with a slight delay from QUEST, and only occurring every second ping cycle due to the echo repeater technique used for these runs. A delay had to be introduced in the echo repeater so that the low latency echo repeat would not coincide with the echo from QUEST. The traces in Fig. 2 were used to identify automatic detections that corresponded to echoes from QUEST and the echo repeater. Once identified, 1 s time-series snippets of un-enveloped, matched-filtered data were extracted for analysis using the classifier. Those echoes not associated with QUEST or the echo repeater were considered to be clutter; however, it is likely that echoes from vessels similar to QUEST were included in the clutter database which could reduce target-clutter discrimination. For example, in Fig. 2 other traces of moving objects can be observed, some of which likely correspond to other vessels and are captured during automatic detection.

![Fig. 2: First half of the PAS run (a) and CAS run (b). The solid line is formed by echoes from QUEST and the dashed line is formed by the echo repeater signals, which were transmitted every second ping with the echo repeater technique used during these runs. Only half of the PRI is shown on the vertical axis.](image-url)
In total, 120 PAS echoes were obtained from QUEST, with a mean SNR of 16.2 dB and standard deviation of 5.0 dB, with all statistics calculated from decibel values. There were 117 CAS echoes from QUEST with a mean SNR of 13.9 dB and standard deviation of 4.3 dB. In addition to the echoes from QUEST and the echo repeater, approximately 175,000 clutter detections were obtained with the detection threshold of 10 dB used. Most of these were not considered in this paper, as will be discussed in the next section. Detection was performed on enveloped matched-filter output, and these detections formed the centre of 1 s snippets extracted from un-enveloped matched-filter output. The snippets containing QUEST echoes formed the target database for training the aural classifier.

3. AURAL CLASSIFICATION RESULTS

3.1. Background

The aural classifier mimics the human auditory system by conditioning echo signals in a similar manner as the outer and inner human ear, and by simulating the cognitive process through representation of the echoes as perceptual features that are used by a Gaussian classifier to determine whether an echo should be designated as a target or clutter. Details on the aural classifier and the aural features it uses are published in [4], [5], and [6].

The classifier is trained by selecting a database of target and clutter echoes to form the training dataset. The training process uses discriminant analysis to formulate a combination of aural features that optimizes separation of the target and clutter echoes in the training dataset. The statistics of the training echoes determine how echoes in the testing dataset are classified. The testing dataset is typically independent of the training dataset to fully evaluate the classifier; however, in this paper, the same training data was used for testing. This removed the requirement to split the limited number of target echoes in the datasets into training and testing portions, which improved training statistics, and resulted in a measure of classification performance that represented the expected maximum achievable with that training configuration.

Previous work verified that the aural classifier performs better with signals of higher SNR [2], as is generally expected for signal classification. Given this SNR dependence, it was important to match the SNR distributions of the target and clutter classes to ensure that discrimination between classes was due to signal features and not a consistent difference in noise background. This was accomplished by forming SNR histograms for each class and removing echoes such that the histogram bin counts matched within 40%. Allowing some discrepancy in the SNR of the two classes allowed more echoes to be included, which improved statistics.

As in previous work [4,5,6], the area under the receiver-operating characteristic (ROC) curve (AUC) was used to evaluate classifier performance.

3.2. Classification Results

After matching the SNR distributions of the target and clutter training sets to within a reasonable approximation, the PAS dataset consisted of all 120 echoes from QUEST and 192 clutter echoes. This data was used to train the classifier and generate a training ROC curve, shown in Fig. 3(a). The AUC value for the curve was 0.88, with probability of
detection, PD = 0.86, and probability of false alarm, PFA = 0.28 at the minimum-error-rate operating point [7]. AUC values above 0.8 indicate excellent discrimination [8].

The procedure was repeated for the CAS data, forming a training dataset with the 117 QUEST echoes and 184 clutter echoes. The resulting ROC curve shown in Fig. 3(b) has AUC = 0.94, indicating outstanding discrimination [8], with PD = 0.87 and PFA = 0.13 at the operating point.

Given that previous experimental validation used 2800 Hz of bandwidth (600–3400 Hz) [5], the results obtained using 900 Hz of bandwidth (1800–2700 Hz) are very promising. The interesting result of higher performance with CAS than PAS was unexpected and warranted preliminary analysis.

![ROC curve for the QUEST/clutter training dataset for the PAS run (a) and the CAS run (b).](image)

### 3.3. Analysis of echoes

When using the aural classifier, preliminary analysis usually involves listening to the echo snippets to see if there are any obvious aural characteristics that could affect classification. In this case, the CAS echoes from QUEST had a distinct chirp sound, compared to broadband impulsive sound of the PAS echoes from QUEST, which sounded similar to PAS echoes previously examined by the authors. A spectrogram of a CAS echo from QUEST, extracted from un-enveloped, matched-filter output, is shown in Fig. 4(d), and is centred on the peak sample within the echo. The chirp effect can be observed in the QUEST echo, and even more clearly on the slightly delayed echo repeat, which starts just after 0.6 s. This phenomenon was not observed in PAS echoes, or in CAS echoes from stationary objects. It was therefore deduced that the effect is due to Doppler mismatch between the replica and Doppler distorted echo from QUEST, and that the effect is dependent on pulse length.

To further investigate the chirp phenomenon, the CAS replica was dilated in time to model 5 kn Doppler shift distortion. This distorted signal was then cross-correlated with the original CAS replica to observe the mismatch effect. A spectrogram of the cross-correlation is shown in Fig. 4(b), and is centred on the peak sample. The chirp caused by the mismatch is clearly visible, and closely resembles the experimental CAS matched-filter output displayed immediately below it in Fig. 4(d). The sweep rate estimated from the spectrogram is 8.0 Hz/ms.
The same procedure was followed to examine if the effect of Doppler mismatch could be observed for the PAS case. The spectrogram of the PAS cross-correlation is shown in Fig. 4(a) with an example of a PAS echo from QUEST shown immediately below it in Fig. 4(c). The chirp effect is not observable on the scale shown in Fig. 4, which was chosen to match the CAS example to allow comparison; however, it is present and could be observed when closely zoomed on the PAS cross-correlation spectrogram with a compressed vertical axis. The estimated sweep rate is approximately 220 Hz/ms.

As the chirp effect is clearly audible, it can presumably affect the aural features, and it is therefore possible that the aural classifier is cueing on this signal feature, resulting in increased discrimination between QUEST echoes and clutter for the CAS case.

4. CONCLUSIONS AND FUTURE WORK

In this paper, we observed that CAS, when processed identically to PAS using a full-band replica, produced echoes with comparable SNR to PAS echoes, even in the complex environment of 20 m littoral waters where coherence loss is expected to have the largest impact.

Discrimination between echoes from the hull of CFAV QUEST and clutter using automatic aural classification was possible using 900 Hz bandwidth, where experimental validation had previously been performed using a much higher bandwidth of 2800 Hz. Discrimination between QUEST and clutter echoes was observed to be higher with CAS than PAS. It is speculated that the aural classifier was cueing on features related to the chirp structure of CAS echoes, which was caused by Doppler shift distortion. The chirp phenomenon was clearly audible and consistent over the entire run because of QUEST’s constant speed and heading. As observed in this paper, this phenomenon could be useful for discriminating moving targets from clutter.
The results presented in this paper are preliminary and further analysis is required to draw conclusions on the classification performance expected for CAS echoes compared to that expected for PAS echoes. Due to the heavy marine traffic during the trial, the PAS and CAS clutter datasets potentially contain different numbers of echoes from other surface ships in the area. High SNR clutter contacts were used to match the SNR distribution of QUEST echoes, which increases the likelihood that echoes from passing ships were in fact included in these clutter databases. This would likely have a large impact on classification because the echo features from other vessels would presumably resemble those of QUEST. In future analysis of the TREX13 data, AIS data and clutter-mitigation images will be used to identify echoes from moving vessels, which will avoid their inclusion in clutter datasets.

5. ACKNOWLEDGEMENTS

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REFERENCES


SIGNAL CHANGE DETECTION METHOD USED FOR MINE-LIKE OBJECTS SEGMENTATION IN SONAR IMAGES

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\textbf{Abstract:} In this paper a Signal Change Detection (SCD) method is used and a novel segmentation (SCDS) approach is presented. SCDS is used for segmenting underwater Mine-Like Objects in side-scan sonar images. Generally, SCD is a statistical method used for detecting the time instances where the amplitude levels of amplitude discrete signals change significantly. SCD approximates 1D real signals by choosing a predefined number of jumps of an amplitude discrete signal that achieves maximum similarity, i.e. fitting probability. Changes may occur at unknown time instances with unknown amplitude levels. In this paper, the row and/or column vectors of 2D Sonar intensity images are considered as 1D signals. Two unknown time instances are detected with three amplitude levels, which are calculated as mean values within the detected time indices. Experimental results show that the SCDS approach achieves an efficient segmentation of sonar images in the desired background, shadow (low intensity) and object (high intensity) region.

\textbf{Keywords:} Mine-Like Objects, Side-scan Sonar, Signal Change Detection, Thresholding, Image Processing.
1. INTRODUCTION

Image segmentation methods are used in image processing for automated object segmentation from its surrounding background. In line with the development of computers, segmentation methods are increasingly used in medical image processing, computer vision, face, iris and fingerprint recognition. Segmentation is typically used in systems where human vision can be replaced with computer vision. In this paper, image segmentation is applied on automated underwater Mine-Like Objects (MLO)detection.

After the World War II, underwater mines pose a threat to humanity, threatening waterways involving ships and submarines. There are several types of underwater mines retained for sinking ships and submarines. In this paper we discussed the three most common underwater mines being laid on the seabed: truncated cone mine, stone mine and cylinder mine.

Underwater mines can be identified by examining the images obtained from a sonar system. Detection of underwater MLO is commonly performed using side-scan sonar moving beneath the water's surface [1]. Such a system, typically mounted on a Remotely Operating or Autonomous Underwater Vehicle (ROV or AUV) is deployed relatively close to the seabed. After completion of the scanning a seabed image is created. To each pixel an 8-bit image intensity is assigned. The recorded images are relatively large and by applying efficient pre-processing method as proposed in [2] one can extract regions of interest (ROI) including a single MLO, c.f. Fig. 1a. The sonar has a view from the bottom to the top of the image. Therefore, an object region appears first (intensity significantly above average), followed by a shadow region (intensity well below average). MLO segmentation result is illustrated in Fig. 1b, where object, shadow and background regions are illustrated with red, blue and green colour, respectively.

Main goal of this paper is to segment MLO by employing a statistical Signal Change Detection (SCD) method, which is presented in Section 2. Signal Change Detection Segmentation (SCDS) method is a novel method for MLO segmentation and is presented in Section 3. Finally, Section 4 concludes this paper.

\[\text{Fig. 1: Region of interest (ROI) image with a Mine-Like Object (MLO) (a); Ground truth MLO segmentation result (b).}\]
2. SIGNAL CHANGE DETECTION

Signal Change Detection (SCD) is a statistical method which can be used for the detection of $W$ changes in a 1D signal $x$, i.e. $x(n), n = 1, ..., N$. The SCD’s output is a list of indices $(n_1, n_2, ..., n_W)$, which enables the approximation of the measured signal $x$ by a signal $\tilde{x}$ with only $W + 1$ distinct amplitudes $(A_1, A_2, ..., A_{W+1})$. The approach is applicable for known or unknown amplitude changes $\Delta A = A_{w+1} - A_w$, which may occur at known or unknown indices $n_w, w = 1, ..., W$ [3]. In this work the SCD method has specifically $W = 2$ changes. It requires two amplitude changes $\Delta A$ at the unknown indices $n_1$ and $n_2$. The amplitude changes and indices are unknown. Hence, SCD can be considered as an optimization problem that provides the indices $n_1 \in [1, n_2 - 1]$ and $n_2 \in [n_1 + 1, N - 1]$ such that $\tilde{x}$ approximates $x$ with maximum probability. The approximating signal $\tilde{x}$ is defined by eq. (1) and it has three amplitude levels $A_1$, $A_2$ and $A_3$ which are defined by the mean values of $x$, before, between and after the indices $n_1$ and $n_2$, c.f. eq. (2).

\[
\tilde{x}(n) = \begin{cases} 
A_1, & n = 1, ..., n_1 \\
A_2, & n = n_1 + 1, ..., n_2 \\
A_3, & n = n_2 + 1, ..., N 
\end{cases} 
\]  

(1)

\[
A_1 = \frac{1}{n_1} \sum_{n=1}^{n_1} x(n), \quad A_2 = \frac{1}{n_2 - n_1} \sum_{n=n_1+1}^{n_2} x(n), \quad A_3 = \frac{1}{N - n_2} \sum_{n=n_2+1}^{N} x(n) 
\]  

(2)

Let $x = \tilde{x} + u$, where $u$ represents additive Gaussian white noise with variance $\sigma^2$. Thus, the probability density function of the measurements can be expressed by eq. (3).

\[
p(x; n_1, n_2) = \frac{1}{(2\pi\sigma^2)^{n/2}} \exp \left[ -\frac{1}{2\sigma^2} \left( \sum_{n=1}^{n_1} (x(n) - A_1)^2 + \sum_{n=n_1+1}^{n_2} (x(n) - A_2)^2 + \sum_{n=n_2+1}^{N} (x(n) - A_3)^2 \right) \right] 
\]  

(3)

The optimization problem of finding the indices $n_1$ and $n_2$ for which $p(x; n_1, n_2)$ takes its maximum value, can be rewritten as a problem of finding the indices $n_1$ and $n_2$ for which eq. (4) takes its minimum value, c.f. eq. (5).

\[
V_{SCD}(n_1, n_2) = \sum_{n=1}^{n_1} (x(n) - A_1)^2 + \sum_{n=n_1+1}^{n_2} (x(n) - A_2)^2 + \sum_{n=n_2+1}^{N} (x(n) - A_3)^2 
\]  

(4)

\[
V_{SCD}(\bar{n}_1, \bar{n}_2) = \min_{1 < n_1 < n_2 < N} V_{SCD}(n_1, n_2) 
\]  

(5)

For an arbitrary large number of jumps $W$, a general solution of eq. (5) can be obtained by applying dynamic programming and integral images [3]. In this work, where $W = 2$, the direct solution is used which considers all combinations of $n$ such that $1 < n_1 < n_2 < N$. Subsequently, the minimization problem defined with eq. (5) has the time complexity of $O(N^2/2)$. In order to speed-up calculation, the direct solution is used in conjunction with the integral image method for mean value calculation, c.f. Algorithm 1.
Algorithm 1. Direct solution for Signal Change Detection (SCD) and two amplitude jumps \((W = 2)\).

Input: \(x\)

Outputs: \(\tilde{n}_1, \tilde{n}_2, A_1, A_2, A_3\)

1: \(\text{Set } y\text{ as 1D integral image of } x\)

2: \textbf{For each} \(n_1 = 1\) to \(n_1 = N - 2\)

3: \(A_1 = y(n_1 + 1)/n_1\)

4: \(S_1 = \sum_{i=1}^{n_1} (x(i) - A_1)^2\)

5: \textbf{For each} \(n_2 = n_1 + 1\) to \(n_2 = N - 1\)

6: \(A_2 = (y(n_2 + 1) - y(n_1 + 1))/(n_2 - n_1)\)

7: \(S_2 = \sum_{i=n_1+1}^{n_2} (x(i) - A_2)^2\)

8: \(A_3 = (y(N + 1) - y(n_2 + 1))/(N - n_2)\)

9: \(S_3 = \sum_{i=n_2+1}^{N} (x(i) - A_3)^2\)

10: \(V_{\text{SCD}} = S_1 + S_2 + S_3\)

11: \textbf{If } \(V_{\text{SCD}} > V_{\text{SCD, max}}\)

12: \(\tilde{n}_1 = n_1, \quad \tilde{n}_2 = n_2\)

13: \(A_1 = y(\tilde{n}_1 + 1)/\tilde{n}_1\)

14: \(A_2 = (y(\tilde{n}_2 + 1) - y(\tilde{n}_1 + 1))/(\tilde{n}_2 - \tilde{n}_1)\)

15: \(A_3 = (y(N + 1) - y(\tilde{n}_2 + 1))/(N - \tilde{n}_2)\)

3. SIGNAL CHANGE DETECTION BASED SEGMENTATION (SCDS)

Signal Change Detection based Segmentation (SCDS) uses SCD for MLO segmentation. Input image \(I_M\) has typically 100 \(\times\) 100 pixels. Before applying image segmentation, \(I'_M\) is created by image pre-processing, c.f. Fig. 2. \(I'_M\) is oriented, according to the sonar point-of-view so that the object region appears on the bottom of the image before the shadow region. Subsequently, \(I'_M\) is suitably padded around its edges by \(k\) pixels. Furthermore, it is filtered by a \((2k + 1) \times (2k + 1)\) median filter which preserves lines and reduces image noise. After the image pre-processing step, image is segmented in four steps. In the first step an image thresholding technique is used for creating the \(I'_0\) and \(I'_S\) images, c.f. Fig. 1. The second step uses the SCD method, c.f. Section 2, for segmentation of object and shadow regions. SCD segmentation is applied to 1D signals that are formed by the column/row pixel intensity values of an image. SCD detects the start and the end of object and shadow regions, c.f. Fig. 3. Vertical SCD (SCD\(_V\)) is applied along image columns in order to create object \(I'_0^V\) and shadow \(I'_S^V\) segmented images. Horizontal SCD (SCD\(_H\)) is applied along image rows and object \(I'_0^H\) segmented image is created. In the third step, the results from the previous steps are combined and the images \(I'_0\) and \(I'_S\) are obtained by applying eq. (6) and (7). Finally, morphological operations [4] are used in the fourth step in order to create object \(I_0\) and shadow \(I_S\) segmented images, c.f. Fig. 2. Examples of proposed intermediate images are illustrated in Fig. 4 and Fig. 5.

![Fig. 2: Signal Change Detection Segmentation (SCDS) steps.](image-url)
Fig. 3: Signal Change Detection used for object (red line) and shadow (blue line) region detection along 1D signal of image column intensities (gray line).

\[ I_0(i,j) = \begin{cases} I_M(i,j), & I_0^{\prime}(i,j) \neq 255 \land I_0^{\prime\prime}(i,j) \neq 255 \\ 255, & I_0^{\prime}(i,j) = 255 \lor I_0^{\prime\prime}(i,j) = 255 \end{cases} \] (6)

\[ I_5(i,j) = \begin{cases} I_M(i,j), & I_5^{\prime}(i,j) \neq 255 \land I_5^{\prime\prime}(i,j) \neq 255 \\ 255, & I_5^{\prime}(i,j) = 255 \lor I_5^{\prime\prime}(i,j) = 255 \end{cases} \] (7)

Fig. 4: Images \( I_0 \) (a) and \( I_5 \) (b) calculated with image thresholding technique, and images \( I_0 \) (c) and \( I_5 \) (d) calculated with \( SCD_y \).
4. DISCUSSION AND CONCLUSION

In this paper a novel segmentation method is presented for MLO image segmentation. The proposed SCDS method is based on an image column/row processing in order to detect two jumps in a 1D signal. SCDS fuses results from thresholding and the SCD method. Finally, morphological operations are used to create homogenous object and shadow segmented regions. The proposed method is implemented in the MATLAB programming language. Exemplarily, MLO images with $100 \times 100$ pixels are considered. The promising segmentation results obtained with rather short execution times recommend the SCDS method as an integral part of an Automatic Target Recognition (ATR) system.

REFERENCES


PASSIVE SONAR DE-NOISING FOR DIVER DETECTION IN PRESENCE OF SNAPPING SHRIMP

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Abstract: Since 2010, TNO has conducted a number of experiments on passive diver detection in Dutch waters. We have demonstrated detection, localization and tracking of divers wearing open circuit and closed circuit underwater breathing apparatus as well as boats. However, until now, we have not conducted any such experiments in warm waters. The soundscape in warm, coastal waters can be very different due to complex bathymetry profile, the presence of breaking waves and biological noise.
In collaboration with Aruba Ports Authority, TNO conducted a series of measurements to evaluate the feasibility of passive diver detection and harbor protection in such an environment. Although the ambient noise level recorded is lower than in the Netherlands, it is dominated by snapping shrimp noise. These snaps constitute a non-Gaussian background noise, for which conventional signal processing techniques are not suited.
In this paper, the effect of the snapping shrimp noise on the correlation, as well as the effect of de-noising techniques applied in the temporal and in the delay domain are demonstrated.

Keywords: Passive sonar, snapping shrimp, de-noising.
INTRODUCTION

Open circuit divers produce acoustic emission due to the turbulent decompression of breathing gas [1]. This emission is very broadband (bandwidth > 200 kHz) and is repeated at the rate of breathing (see Fig. 1(a)). The diver signal is received on separated hydrophones with a delay that is function of the direction of the source with respect to the hydrophones (Fig. 1(b)).

The generalized cross-correlation (GCC) [2] is a mathematical tool to measure the delay between signals. The GCC of the signals $x(t)$ and $y(t)$ parameterized by $\Psi(\omega)$ is computed as the inverse Fourier transform $F^{-1}$ of the cross-spectrum $C_{x,y}(\omega)$ weighted by the function $\Psi(\omega)$:

$$GCC_{x(t),y(t)}(\tau) = F^{-1}\{\Psi(\omega)C_{x,y}(\omega)\}$$

The GCC reduces to the conventional cross-correlation when $\Psi(\omega) = 1$. The GCC is especially suited to broadband signals, such as the diver’s signal. Computation of the GCC over successive frames of signal enables building a picture that is function of time and delay. Fig. 1(c) presents a correlogram showing a dotted track corresponding to a diver’s inhalations and continuous tracks associated with boats. The delay of the detected sources can be converted to direction. Detections from several pairs of sensors can be combined for localization of the detected sources. Using this approach, detection and localization of divers and small boats was demonstrated experimentally [3].

Snapping shrimp are little shrimp (few centime size) that use their claw to make a cavitation bubble for hunting and communication purposes [4]. The bubble collapse generates a broadband, transient emission. Snapping shrimp are found in nooks and crannies (e.g. rocky bottom) in warm, shallow waters. They snap day and night. Their presence leads to a background of impulsive noise coming from random directions that is neither stationary nor Gaussian. Fig. 2(a) presents one second of acoustic signal recorded

SNAPPING SHRIMP NOISE AND ITS EFFECT ON PASSIVE DETECTION

Snapping shrimp are little shrimp (few centime size) that use their claw to make a cavitation bubble for hunting and communication purposes [4]. The bubble collapse generates a broadband, transient emission. Snapping shrimp are found in nooks and crannies (e.g. rocky bottom) in warm, shallow waters. They snap day and night. Their presence leads to a background of impulsive noise coming from random directions that is neither stationary nor Gaussian. Fig. 2(a) presents one second of acoustic signal recorded
in the port of Oranjestad in Aruba in December 2013. Hundreds of snaps are present at each second; many of which are clearly visible in the time signal. Fig. 2(b) shows a close-up view on individual snaps, illustrating their very short duration and broadband character.

![Fig.2: (a) Signal with snaps from shrimps, (b) close-up view on individual snaps.](image)

GCC enables to detect the strongest source in a given frequency band. Snaps, which are broadband and loud, result in the occurrence of transient peaks in the GCC at random delays, as illustrated in Fig. 4(a).

**DENOISING METHODS**

1.1. **Snap blanking**

The first proposed method for snap de-noising consists in detecting and removing the snaps in the time domain. Snaps are detected as local extrema whose absolute value exceed the threshold \( m + \text{thres} \times \sigma \), where \( m \) and \( \sigma \) are the mean and standard deviation of the absolute value of the signal. The detected snaps are then removed by blanking, i.e. setting the samples to zeros in a vicinity of the detections. In the presented examples, \( \text{thres} \) is set to 3.5 and 22 samples centered on the detected snaps are blanked. This procedure is illustrated in Fig. 3(a) and (b), which show an experimental signal before and after snap blanking, respectively. Note that the remaining samples over the threshold after snap removal are due to transients whose duration exceeds the blanking window duration.

1.2. **Cross-spectrum snapshot selection**

The cross-spectrum \( C_{x,y}(\omega) \) in equation (1) can be obtained as an average over snapshots: \( C_{x,y}(\omega) = \left< C_{x,y}^k(\omega) \right>_{k \in \{1..n_s\}} \), where \( C_{x,y}^k(\omega) \) is the \( k \)th cross-spectral snapshot. Note that a snapshot corresponds to a time window and is unrelated to snapping shrimp. Accordingly, a snapshot does not necessarily include snaps from shrimp.

While the source of interest is stationary, the snapshot \( C_{x,y}^k(\omega) \) is representative of the source in absence of snap; it is corrupted by shrimp noise in presence of snap. In the averaging, the stationary contribution of the source is enhanced by coherent summation while the contribution from the snaps is reduced by the incoherent summation resulting from their random character. The signal to noise ratio (SNR) is therefore improved by increasing the averaging time. It is however limited by the duration during which the signal of interest can be considered stationary. In particular, increasing the averaging
duration past the signal duration (e.g. duration of an inhalation) does not lead to further increase in SNR.

**Fig. 3:** Illustration snap blanking (a,b), cross-spectra snapshot selection (c,d,e) and MIN filtering (f,g). See text for details.

Intuitively, it seems a cross-spectrum estimate that is most representative to the source of interest could be obtained by limiting the summation to the snapshots that do not include snaps. Application of this idea requires a method to separate the *good snapshots*, which are snap free, from the *bad snapshots* corrupted by shrimp noise. A key property to achieve this separation is the fact that the cross-spectra of the good snapshots are similar to each other (because the source is stationary), while the bad snapshots are different from each other (because the snaps come from random directions). We propose to compute the covariance matrix $\eta_{k,t}$ of the cross-spectral snapshots:

$$
\eta_{k,t} = \frac{\int c_{k,x}^k(\omega)c_{l,y}^k(\omega)d\omega}{\sqrt{\int c_{k,x}^k(\omega)c_{k,x}^k(\omega)d\omega \int c_{l,y}^l(\omega)c_{l,y}^l(\omega)d\omega}}
$$

An example of the real part of the snapshot cross-spectra covariance matrix is shown in Fig. 3(c). The average value was computed for each line and is shown in Fig. 3(d) after sorting them in decreasing order. In this representation, the good snapshots have low sorted snapshot index while the bad snapshots have high sorted snapshot index. The limit between the two classes is selected as the sorted index for which the distance to the line joining the first and last point is maximized. It is shown by a red line in Fig. 3(d). Fig. 3(e) shows the covariance matrix as a function of the sorted snapshot index. The lower left part delimited by the red line corresponds to the part associated with the good snapshots. Once
the good snapshots are determined, the GCC can be computed from (1) using for the cross-spectrum:

$$C_{x,y}(\omega) = \langle C_{x,y}^{k}(\omega) \rangle_{k \in \text{good snapshots}}$$

(3)

1.3. MIN filter

The third method considered is a post-processing applied to the correlogram. Given an original correlogram where $X[t_k, \tau_m]$ is the value of GCC at time $t_k$ and delay $\tau_m$, the filtered correlogram $Y[t_k, \tau_m]$ is obtained as:

$$Y[t_k, \tau_m] = \text{MIN}\{X[t_{k-2}, \tau_m], X[t_{k-1}, \tau_m], X[t_k, \tau_m]\}$$

(4)

When a signal is stationary, three successive values of the original correlogram $X$ are similar. The corresponding value of the filtered $Y$ is therefore also a similar value: the signal is preserved, as illustrated in Fig. 3(f). A transient that is present only at a particular instant, however, will be removed, as illustrated in Fig. 3(g).

4. RESULTS

Fig. 4 presents the effect of the proposed processing on 2 minutes of experimental signal collected in the port of Oranjestad in Aruba in December 2013. The signal is sampled at about 200 kHz and is processed using frames of $2^{17} = 131072$ samples with 50% overlap. Each frame is divided into $2^{10} = 1024$ samples snapshots without overlap. Fig. 4(a) shows that direct application of GCC results in a very noisy picture. Fig. 4(b,c,d) demonstrate substantial improvement resulting from the application of the cross-spectral snapshot selection (smart GCC in the figure, Fig. 4(b)), the blanking procedure (c) or both (d). The background of spurious peaks is significantly reduced. The masking of components of interest by snaps is reduced, which is illustrated, for instance, by the fact boat tracks are more consistently populated instead of presenting many discontinuities at the time of strong snaps.

The last row of Fig. 4 illustrates how the MIN filtering removes most of the remaining transients from the correlogram. It also reduces the apparent duration of the diver’s inhalations. Contrary to the other processing methods, which achieved increased SNR by both enhancing the signal and reducing the noise, the MIN filter increases the SNR by reducing the signal less than the noise.

5. CONCLUSION

The problem of detecting stationary sound sources using generalized cross correlation (GCC) in the presence of impulsive noise was considered. Since GCC enables detection of the loudest source in a given frequency band, the presence of loud impulsive noise can mask relatively weaker stationary sources, making their detection challenging.

Three methods have been introduced to limit the influence of the impulsive noise: blanking the snaps in the time domains (pre-processing), selecting cross-spectral snapshots
that are snap-free (enhanced processing), and image processing of the correlogram using the MIN filter (post-processing).

The effectiveness of these methods is shown using experimental data. The detectability using GCC of scuba divers and boats in presence of snapping shrimp is demonstrated.

![Image of correlograms with various processing methods]

**Fig.4:** Effect of the proposed methods on the correlogram of experimental data including divers, boats and snapping shrimp. (a) GCC only, smart GCC (snapshot selection) in the second column (b, d, f), snap blanking on the second row (c, d), snap blanking and MIN filter (keeping the lowest of 3) on the third row (e, f).

**REFERENCES**


TARGET DOPPLER ESTIMATION AND RANGE BIAS COMPENSATION USING LFM HIGH DUTY CYCLE SONAR

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Abstract: Unlike conventional Pulsed Active Sonar (PAS) which listens for echoes in between short-burst transmissions, High Duty Cycle (HDC) sonar attempts to detect echoes amidst the continual interference from source(s) transmitting with nearly 100% duty cycle. The potential advantage of HDC is an increased number of continuous detection opportunities, leading to improved target detection, localization, tracking, and classification. Continuous transmission waveforms may be of several types. Linear frequency modulated waveforms (LFMs) provide good range measurements but not target Doppler; continuous waveforms (CWs) provide good Doppler measurements but not target range. This paper describes an algorithm for Doppler estimation within a single cycle of a continuous LFM HDC signal. Using this method, target Doppler can be estimated more quickly than the typical approach of estimating range-rate over multiple ping cycles. This approach is possible when the HDC LFM signals are processed over short time intervals, which provides a set of multiple measurements within each waveform cycle. The Doppler estimate is provided at the information processing stage, and therefore is available to the tracker for improved target state estimation. Additionally, the obtained Doppler estimates can be used to correct the measurements’ range bias errors which are inherent in processing LFM signals. Bias errors are a more significant problem for HDC sonar than for PAS because of the reduction in transmitted bandwidth per unit time (frequency sweep rate). They may pose a particular problem in sensor fusion between multistatic sonar sensors by preventing data association (gating). However, these bias errors can be determined using the estimated Doppler and compensated for. The Doppler estimation and bias correction algorithm is applied to data from the TREX’13 experiment, and shows the effectiveness of the method.

Keywords: Continuous active sonar, localization, bias correction, Doppler estimation
1. INTRODUCTION

Recently, there has been emerging interest in the concept of High Duty Cycle (HDC) Sonar. Unlike Pulsed Active Sonar (PAS) which listens for echoes in between short transmission bursts, HDC sonar attempts to detect echoes amidst the continual interference from source(s) transmitting with nearly 100% duty cycle. A schematic of the two contrasting approaches is shown in Fig. 1. Unlike a PAS system, HDC can provide multiple detection opportunities within a single FM ping repetition interval or cycle ($T_{pri}$) when processed appropriately. Less time lapse between detections allows near continuous holding on a target and constrains the growth of its area of uncertainty (AOU) [1]. This will improve target localization, holding, maneuver detection, and false track rate.

This paper describes a method utilizing LFM HDC signals to quickly ascertain target range-rate as well as correct for range-bias errors which are inherent in LFM signal processing. Whereas PAS LFM signals often yield negligible bias errors, HDC LFM signals will have one order of magnitude larger bias errors than PAS signals due to reduced frequency sweep rate. This results in target mis-localization which degrades target tracking and multi-sensor fusion. Furthermore, systems using FM signals obtain target velocity estimates only through a multi-ping observation of the change in target range (range-rate). In PAS systems, the measurement update interval corresponds to the ping repetition interval, so range-rate estimation may take some time before it is obtained. With HDC LFM signals, the update rate is much quicker, with many potential target observations within a single $T_{pri}$ (intra-cyclical). The intra-cyclical observations can be used to estimate the range-rate in a much shorter time compared to a PAS system.

2. HIGH DUTY CYCLE SONAR USING LFM SIGNALS

An LFM signal’s instantaneous frequency is "swept" linearly over time. It is the derivative of the signal’s phase term, and is given by

$$f = f_i + kt; \quad k = \frac{f_2 - f_1}{T_{pri}}. \quad (1)$$

where $k$ is the waveform frequency-time slope (or sweep rate), $T$ is the LFM signal

Fig. 1. Depiction of three cycles of PAS (top left, blue) and HDC (bottom left, red) transmission methods. Processing of an HDC LFM signal (right).
duration, \( f_1 \) is the sweep start frequency, \( f_2 \) is the sweep stop frequency.

The use of continuous repeating linear FM (LFM) signals in HDC provides the ability to obtain range estimates of detected targets by measuring echo time delays. Fig. 1 shows an LFM waveform with total bandwidth, \( B \), and a duration, \( T \), which is the same as its ping repetition interval, \( T_{pri} \). An echo from a target arrives with time delay, \( \Delta \tau \). This is obtained by heterodyning (de-chirping) the received signal and performing spectral processing [2]. The delay is obtained from the frequency shift as

\[
\tau = \Delta f \cdot \frac{T_{pri}}{B}.
\]

The signal ambiguity function provides information about the sensitivity and accuracy of a waveform to target Doppler and delay time (which corresponds to range). The LFM signal’s ambiguity function is given by [3]

\[
|X_{FM}(f, \tau)|^2 = \left( 1 - \frac{|\tau|}{T} \right)^2 \left\{ \frac{\sin[\pi(f-kr)(T-|\tau|)]}{\pi(f-kr)(T-|\tau|)} \right\}^2
\]

where \( \tau \) is the time delay, and \( |\tau| \leq T \). An example LFM signal’s ambiguity surface is shown in Fig. 2 (left). The image shows that when a target has Doppler, the processing will yield a bias error in the measured time delay. In this example, a 10 kt target will have a time delay bias error of about 0.3 seconds.

The LFM measures time delay and is related to the time bias, which is a function of target Doppler as [4]

\[
\tau_{measured} = \tau_{true} + \tau_{bias}; \quad \tau_{bias} = \frac{2v}{ck} \cdot f_c
\]

\( v \) is the target’s range-rate (in kts), \( c \) is the speed of sound, \( f_c \) is the waveform’s (or processed waveform section’s) center frequency, and where we ignore noise and estimate only the mean values. From this equation we see that the bias error worsens with increasing waveform operating frequency and target Doppler, and decreasing waveform frequency-time slope (i.e., lengthening \( T \) for fixed \( B \) or reducing \( B \) for fixed \( T \)).

Fig. 2 (right) shows an example of the expected target time delay as a function of time, for 3 cycles of an LFM waveform of the type used in subsequent analysis. The target in this case is assumed to be opening range at a speed of 5 kts. The red curve represents the true target range-rate. The dashed blue curve represents the expected measurements of time-delay when each LFM cycle is processed wholly and coherently as one waveform. In this case the processing interval \( (T_p) \) is equal to \( T_{pri} \), and bias error is observed to be constant over time (parallel to truth). However, if the processing is broken down into sub-sections, i.e., \( T_p < T_{pri} \), then the resulting time delay has a “sawtooth” pattern which varies around the previous result. This is shown as the solid blue line, where the signal is broken into 18 smaller processing intervals. This is due to the fact that each processing interval has a slightly different center frequency and thus an increased slope for the bias is obtained within each cycle. It is this saw tooth effect and the measurement of its slope vs. time delay within a HDC cycle that enables the Doppler estimation.

We define the Inter-Ping Slope (IPS) as the slope of the target time-delay from one ping cycle to the next, as follows:

\[
IPS = \frac{\Delta \tau}{\Delta t} = \frac{2\Delta r/c}{T_{pri}} = \frac{2\dot{r} T_{pri}}{c T_{pri}} = \frac{2\dot{r}}{c}
\]

(5)
Now, we define the Intra-Cyclical Slope (ICS) as the slope of the target time-delay within a single ping cycle as:

\[ ICS = \frac{\Delta t}{\Delta \tau} = IPS + \frac{\tau_{bias}(f_2) - \tau_{bias}(f_1)}{T_p} = IPS + \frac{f_2 T_p - f_1 T_p}{T_p} = \frac{4r}{c} \]  

(6)

From these we see that with measurement of either IPS or ICS we can estimate the target range-rate according to

\[ \hat{r} = \frac{cIPS}{2} = \frac{cICS}{4} \]  

(7)

Estimation based on IPS will require more time with multiple ping cycles processed (with resulting detections) and analysed before the target Doppler can be obtained. Using ICS, we have the potential to estimate the target range-rate within a single transmission cycle, which is much quicker than what can be obtained using the IPS measurement. Furthermore, if we treat the multiple measurements within a single cycle as a batch, we can compensate for their bias errors prior to utilizing them in tracking and localization.

3. DOPPLER ESTIMATION AND RANGE BIAS CORRECTION ALGORITHM

We now describe a practical processing method to obtain the ICS, and then use the resulting Doppler estimate to correct for the time bias errors. We assume that the data will have undergone the following standard HDC processing chain: beamforming, de-chirp heterodyning, low-pass filtering, spectrogram processing, normalization, detection peak clustering and extraction (above some detection threshold). Range is related to time delay according to: \( r = c\tau/2 \).

We assume that the HDC processing uses a processing interval for the STFT which is short compared to the ping duration \( (T_p < T_{pr.i}) \), resulting in multiple detection opportunities, \( N = T/T_p \), per waveform cycle. The processed detection contacts from one cycle are considered as batch measurements for which the ICS will be extracted. There may be multiple targets and fewer than \( N \) detections per target in each batch.

The ICS is determined by means of the Hough transform [5], which is a common image processing method for detecting lines (and other) shapes within images. The Hough transform uses a parameter space representation of a line using the polar form as

\[ \rho = \tau \cdot \cos \theta + t \cdot \sin \theta, \]  

(8)
where $t$ and $\tau$ represent our quantities of (slow) ping time and delay time, respectively. Using such a representation, input image lines map to single points. Each point in the input image $(\tau, t)$ is thus mapped into an accumulator array in the $(\rho, \theta)$ parameter space, with each pixel containing the count of input points contributing to it. A threshold is then set on the output accumulator array requiring $M$ out of $N$ detection counts; otherwise the detection cluster is removed from further processing. Where the $M$ of $N$ rule is satisfied, the peaks of the Hough output image in $(\rho, \theta)$ are extracted, and treated as detected lines. These can then be converted to slope-intercept form to obtain the ICS.

For each set of contacts, a measurement of ICS is made. The target Doppler is then obtained using Eqn. 7. This range-rate estimate is the best linear fit of the target’s motion over the current cycle being processed. We assume the cycle time is short enough that ASW targets can reasonably be assumed to be traveling according to a nearly constant velocity (NCV) motion model. The resulting target Doppler is now associated with all the contacts which contributed to its estimation, and may be utilized to characterize or classify the detection group. It may also be incorporated as additional useful information into a subsequent target tracking process, though that has not been attempted to date.

Once the target Doppler has been estimated, we can correct the time/range bias errors. This is done by reforming Eqn. 4 as follows

$$
\tau_{\text{compensated}}(t) = \tau_{\text{measured}}(t) - \frac{2\pi}{ck} f_c(t)
$$

(9)

Here we must take into account the fact that $f_c(t)$ steps up with each successive processing interval in time, according to the LFM sweep. All detections within each ping cycle are individually Doppler compensated. The output of this process will remove both the effect of the constant bias offset and the “saw tooth” effect. The resulting contact information is more accurate and sent to a target tracker without biases.

4. APPLICATION TO TREX’13 SEATRIAL DATA

The TREX’13 sea trial [6] was conducted during May, 2013, off the coast of Panama City, Florida. The HDC portion of the experiment utilized a fixed, bottom-moored acoustic source and hydrophone line array, in a nearly monostatic configuration. These were operated by the R.V. SHARP, which was moored nearby and collected the receiver data. The trial was situated in an extremely shallow area with water depths of about 10-12 meters. The target was provided by DRDC-Atlantic, and consisted of the SMART Echo Repeater (E/R) system [6], which was towed by the Canadian vessel R.V. Quest. The track trajectories were run radially inbound and outbound from the monostatic sonar, over ranges of 10-15 km, along two different bearing lines. The SMART E/R system provided a way to simulate an actual target. It has various operating modes, but its fundamental capability is to capture the transmitted signal and retransmit it to simulate an echo.

For this analysis we have focused on Run 63 of the experiment. The source transmitted an upsweep LFM signal of 18 seconds duration over the frequency band of 1800-2700 Hz. The signal was continuously repeated, but with a two-second (non-transmitting) gap between each cycle ($T = 18 \, s, T_{pr} = 20 \, s$). The R.V. Quest maintained an outbound radial trajectory at a speed of 5 kts, with the E/R operating in the “ping-pong” mode. This means that the E/R captured every odd source transmission ping cycle and then
retransmitted it on every even transmitted cycle with an amplified level of 25 dB (target strength). In this manner E/R echoes were obtained on the receiver every other ping cycle with the correct echo timing. Therefore, while the source was transmitting with a 90% duty cycle, the E/R was transmitting with only a 45% duty cycle. So, target echoes were to be expected only every other LFM transmission cycle. Despite this constraint, the data was still very useful in developing HDC processing techniques and analyzing performance of the approach. Run 63 was one hour in duration; we focus our analysis on the first 30 minutes of the run.

The received data were beamformed, although for this analysis, only a single beam was analyzed because the E/R target maintained its trajectory solely within this beam. The data were heterodyned (de-chirped) using the known transmission signal, and low pass filtered. Spectrogram processing was then performed with an FFT integration time ($T_f$) of 1 second. With this processing we expect about 18 detection opportunities on every other ping transmission cycle. Fig. 3 (left) shows the spectrogram output for the entire 30 minutes. The heterodyned direct blast is shown at zero frequency; the target is shown with negative, decreasing frequency corresponding to increasing time delays of the target as it opens in range.

The data were then normalized, and detection contacts were formed. These were sent to the Doppler estimation and range-bias correction algorithm in blocks of twenty scans at a time, (corresponding to $T_{pri}$). Fig. 3 (right) shows the input contacts (in a scaled time delay space) for a single HDC ping cycle, with the resulting Hough estimated line shown overlaid.

Fig. 3. The heterodyned spectrogram output (left); relative acoustic receive levels (in dB) as a function of time and frequency. Example contact group (black markers) and ICS estimate (red line) using the Hough transform (right)

Fig. 4 shows the results of the application of the bias correction algorithm to the entire 30 minutes. The blue markers show all the input detection contacts. We observe the target opening in range from time delays of 1.0 to 6.5 seconds over the 30 minutes. Also present, are various fixed clutter features, though they appear to have fewer detections and less scan-to-scan consistency. The red markers show the output of the algorithm, which are the detection contacts with bias correction for all groups which passed the M/N test. We see that the algorithm also filters out some inconsistent clutter features.
Fig. 4 also shows a detailed view of a section of the output over 5 cycles of transmission. Every other cycle (3 total shown) has a number of detections. The Doppler compensated contacts are shown with smaller time delays (i.e., ranges) than the uncompensated detections. Lines are overlaid to help visualize the slope of each group relative to the others. The input detection groups have the same slope, but don’t fall on the same line as they exhibit the saw tooth pattern. For the output points (in red), the slopes of the individual groups are consistent with the slope from group-to-group. The saw tooth pattern has effectively been removed, and the range-bias appropriately corrected.

Next the data was input into a multi-sensor, multi-target tracker. The E/R system was localized with GPS accuracy for a true reconstruction of target position and speed. Tracker results were obtained for both the input (uncompensated) data and the output (compensated data). Fig. 5 shows the results for segment of the scenario with the true target range overlaid. The uncompensated track is seen offset from the truth due bias errors and includes some jitter due to the “saw tooth” effect. The compensated track shows better localization accuracy and a smoother trajectory estimate.

5. SUMMARY

This paper has described an algorithm for Doppler estimation applied within a single cycle of a continuous HDC LFM signal, using the Hough transform. Using this method, target Doppler can be estimated more quickly than the typical approach of estimating range-rate over multiple ping cycles as would be done with PAS. The Doppler estimate is provided at the information processing stage, and therefore is available to the tracker for improved target state estimation. Additionally, we have shown a method to use the obtained Doppler estimates to correct the measurements’ range bias errors which are inherent in processing LFM signals. The algorithm was successfully applied to sea trial data from the TREX’13 experiment, demonstrating the effectiveness of the algorithm.

6. ACKNOWLEDGEMENTS

We acknowledge Dr. Paul Hines, Mr. Jeff Scrutton, and Mr. Stefan Murphy of Defence R&D Canada-Atlantic who designed and conducted the TREX’13 experiment; Dr. Dajun Tang, Dr. Todd Hefner, and Dr. Kevin Williams of the Applied Physics Laboratory of the University of Washington (APL-UW) who managed and led the TREX’13 trial, and Dr. John Preston of Pennsylvania State University's Applied Research Lab who managed quality control and data collection for the experiment. We wish to acknowledge the officers and crew aboard CFAV QUEST and RV SHARP, and the APL-UW dive team for their support throughout the trial. Funding for this work was provided by ONR Code 32 and ONR Global-London.
Fig. 4. Algorithm input (blue markers); algorithm output (red markers). Run 63 (left) and five cycle segment (right); group slopes are indicated with the lines.

Fig. 5. Estimates of the target range over 100 second period; uncompensated range bias errors (blue), compensated range bias errors (red), and true target range (black).

REFERENCES

TARGET AOU GROWTH CONTAINMENT USING
LFM HIGH DUTY CYCLE SONAR

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Abstract: Unlike conventional Pulsed Active Sonar (PAS) which listens for echoes in between short-burst transmissions, High Duty Cycle (HDC) sonar attempts to detect echoes amidst the continual interference from source(s) transmitting with nearly 100% duty cycle. The potential advantage of HDC is an increased number of continuous detection opportunities, leading to improved target detection, localization, tracking, and classification. A common PAS and HDC sonar transmission waveform is the linear frequency modulated signal (LFM), which provides good target range estimation. The range accuracy of the waveform is inversely proportional to the processed signal bandwidth. In traditional PAS sonar processing, the full signal bandwidth is transmitted over a short time interval within the repeat cycle. It may be processed with a matched filter, resulting in a single detection opportunity. In HDC sonar processing, the signal’s bandwidth is spread out over the entire transmission repeat cycle. It may be split up into sub-bands by processing shorter time blocks, resulting in multiple detection opportunities per waveform cycle. Though the probability of detection and ranging accuracy may be lower for HDC sonar than for PAS (due to reductions in source level and processed bandwidth), a distinct advantage is that there is less time lapse between measurement scans. Such a rapid measurement update-rate effectively contains the growth of the target’s area of uncertainty (AOU) within a kinematic tracker. This may improve target localization, holding, manoeuvre detection, and false track rate. The value of this effect is demonstrated through processing and analysis of HDC sonar data from the TREX’13 at-sea experiment.

Keywords: Continuous active sonar, localization, target tracking, High duty cycle
1. INTRODUCTION

Recently, there has been emerging interest in the concept of High Duty Cycle (HDC) Sonar. Unlike Pulsed Active Sonar (PAS) which listens for echoes in between short transmission bursts, HDC sonar attempts to detect echoes amidst the continual interference from source(s) transmitting with nearly 100% duty cycle. A schematic of the two contrasting approaches is shown in Fig. 1. Unlike a PAS system, HDC can provide multiple detection opportunities within a single FM ping repetition interval or cycle ($T_{pri}$) when processed appropriately. Less time lapse between detections allows near continuous holding on a target and constrains the growth of its area of uncertainty (AOU). This will improve target localization, holding, maneuver detection, and false track rate. In addition, it may provide a method to more quickly ascertain target range-rate as well as correct for LFM range-bias errors [1].

![Fig. 1. Depiction of three cycles of PAS (top left, blue) and HDC (bottom left, red) transmission methods. Processing of an HDC LFM signal (right).](image)

2. CONTINUOUS ACTIVE SONAR USING LFM SIGNALS

The LFM signal is one in which the instantaneous frequency is “swept” linearly over time. The signals can be swept up (“upsweeps”) or down (“downsweeps”). LFM signals are convenient for HDC operations because time delay is straightforwardly extracted from measurements of Doppler shift. The equation for an LFM signal is given by:

$$s_{FM}(t) = \sqrt{2/T} \cdot \cos \left( 2\pi \left( f_1 t + k \frac{t^2}{2} \right) \right); \quad -\frac{T}{2} \leq t \leq \frac{T}{2}.$$  \hspace{1cm} (1)

where the waveform frequency-time slope (or sweep rate) is given by

$$k = \frac{f_2 - f_1}{T},$$  \hspace{1cm} (2)

where $T$ is the LFM signal duration, $f_1$ is the sweep start frequency, $f_2$ is the sweep stop frequency. The instantaneous frequency is given by the derivative of the signal’s phase term:

$$f = f_1 + kt.$$  \hspace{1cm} (3)
The use of a continuous (or near continuous) repeating linear FM (LFM) signal in HDC provides the ability to obtain range estimates of detected targets by measuring echo time delays. Fig. 1 also shows an LFM waveform with total bandwidth, $B$, and a duration, $T$, which is the same as its ping repetition interval, $T_{pri}$. The echo from a target arrives with time delay, $\tau$. This is obtained by heterodyning (de-chirping) the received signal and performing spectral processing [2-3]. The delay is obtained from the frequency shift as

$$\tau = \Delta f \cdot \frac{T_{pri}}{B}. \hspace{1cm} (4)$$

3. TARGET LOCALIZATION

With HDC LFM processing we have the freedom to select the processing time/bandwidth. Reducing the processing bandwidth will decrease the range resolution, which increases the AOU of target echoes on each measurement. And, it may also potentially reduce target echo signal-to-noise ratio. However, at the same time, it will provide increased numbers of detection opportunities per waveform cycle, with less time lapse from scan-to-scan. This will contain the growth of the target AOU. Though the HDC AOU may be larger to start with than its equivalent in PAS, it may be maintained with smaller size through frequent updates whereas the PAS system’s AOU will grow unconstrained until the next measurement is obtained. There will be a tradeoff between achieving reductions in AOU growth and detection resolution, which should be carefully considered. A full analysis of the negative consequences of such processing is not fully addressed here; rather we focus on demonstrating the advantages of HDC in target localization and holding.

We assume the target state vector at time $t_k$, is given by the mean $x$ and $y$ positions and velocities as

$$X_k = [x_k \ y_k \ \dot{x}_k \ \dot{y}_k]^T. \hspace{1cm} (5)$$

We assume some uncertainty in the estimate, which is captured by the target state covariance matrix

$$P_k = \begin{bmatrix} \sigma_x^2 & \sigma_{xy} & \sigma_{xx} & \sigma_{xy} \\ \sigma_{xy} & \sigma_y^2 & \sigma_{xy} & \sigma_{yy} \\ \sigma_{xx} & \sigma_{xy} & \sigma_x^2 & \sigma_{xy} \\ \sigma_{xy} & \sigma_{yy} & \sigma_{xy} & \sigma_y^2 \end{bmatrix}. \hspace{1cm} (6)$$

The target positional error covariance and the target velocity error covariance are given by sub matrix of $P_k$ as

$$P_{pos} = \begin{bmatrix} \sigma_x^2 & \sigma_{xy} \\ \sigma_{xy} & \sigma_y^2 \end{bmatrix}; \quad P_{vel} = \begin{bmatrix} \sigma_x^2 & \sigma_{xy} \\ \sigma_{xy} & \sigma_y^2 \end{bmatrix}. \hspace{1cm} (7)$$

We define the positional and velocity areas of uncertainty (AOU) as
where \( p \) is the desired probability of the target being inside the covariance ellipse. Until another measurement can be obtained and successfully associated with the current estimate, the target state and its uncertainty is projected forward to a future time, \( t_{k+1} \), according to the nearly constant velocity (NCV) target motion model \([4]\) as

\[
X_{k+1} = \Phi_k X_k + w_k, \quad P_{k+1} = \Phi_k P_k \Phi_k^T + Q_k.
\]

where \( \Phi_k \) is the state transition matrix given by

\[
\Phi_k = \begin{bmatrix}
1 & 0 & \Delta t_k & 0 \\
0 & 1 & 0 & \Delta t_k \\
1 & 0 & 1 & 0 \\
0 & 1 & 0 & 1
\end{bmatrix}, \quad \Delta t_k = t_{k+1} - t_k.
\]

\( w_k \) is modeled as a zero-mean white Gaussian process as \( w_k \sim \mathcal{N}(0, Q_k) \), where \( Q_k \) represents the process noise given by

\[
Q_k = \begin{bmatrix}
\frac{1}{2}q_x(\Delta t_k)^2 & 0 & \frac{1}{2}q_x(\Delta t_k)^2 & 0 \\
0 & \frac{1}{2}q_y(\Delta t_k)^2 & 0 & \frac{1}{2}q_y(\Delta t_k)^2 \\
\frac{1}{2}q_x(\Delta t_k)^2 & 0 & q_x \Delta t_k & 0 \\
0 & \frac{1}{2}q_y(\Delta t_k)^2 & 0 & q_y \Delta t_k
\end{bmatrix},
\]

where \( q_x \) and \( q_y \) represent process noise parameters in x and y coordinates, which accommodate for target maneuvers. The process noise parameters can be increased to model a highly maneuverable target. For targets that are not highly evasive and whose motion does not depend significantly on the environment, the NCV model is adequate. As the time between measurement scans, \( \Delta t_k \), increases the process noise term increases, and the localization uncertainty increases. HDC processing enables a quicker update rate, reducing \( \Delta t_k \), and minimizing the growth of the AOU in both position and velocity.

4. ANALYSIS OF THE TREX’13 SEATRIAL DATA

The TREX’13 sea trial \([5]\) was conducted during May, 2013, off the coast of Panama City, Florida. The HDC portion of the experiment utilized a fixed, bottom-moored acoustic source and hydrophone line array, in a nearly monostatic configuration. These were operated by the R.V. SHARP, which was moored nearby and collected the receiver data. The trial was situated in an extremely shallow area with water depths of about 10-12 meters. The target was provided by DRDC-Atlantic, and consisted of the SMART Echo Repeater (E/R) system \([5]\), which was towed by the Canadian vessel R.V. Quest. The track trajectories were run radially inbound and outbound from the monostatic sonar, over ranges of 10-15 km, along two different bearing lines. The SMART E/R system provided a way to simulate an actual target. It has various operating modes, but its fundamental capability is to capture the transmitted signal and retransmit it to simulate an echo.

For this analysis we have focused on Run 63 of the experiment. The source transmitted an upsweep LFM signal of 18 seconds duration over the frequency band of 1800-2700 Hz.
The signal was continuously repeated, but with a two-second (non-transmitting) gap between each cycle ($T = 18\, \text{s}, T_{\text{pri}} = 20\, \text{s}$). The R.V. Quest maintained an outbound radial trajectory at a speed of 5 kts, with the E/R operating in the “ping-pong” mode. This means that the E/R captured every odd source transmission ping cycle and then retransmitted it on every even transmitted cycle with an amplified level of 25 dB (target strength). In this manner E/R echoes were obtained on the receiver every other ping cycle with the correct echo timing. Therefore, while the source was transmitting with a 90% duty cycle, the E/R was transmitting with only a 45% duty cycle. So, target echoes were to be expected only every other LFM transmission cycle. Despite this constraint, the data was still very useful in developing HDC processing techniques and analyzing performance of the approach. Run 63 was one hour in duration; we focus our analysis on the first 30 minutes of the run.

The received data were beamformed, although for this analysis, only a single beam was analyzed because the E/R target maintained its trajectory solely within this beam. The data were heterodyned (de-chirped) using the known transmission signal, and low pass filtered. Spectrogram processing was performed using different processing parameters, as shown in Table 1.

<table>
<thead>
<tr>
<th>Case</th>
<th>Number of sub-bands, $N$</th>
<th>Processing time, $T_p$ (s)</th>
<th>Processing bandwidth, $B_p$ (Hz)</th>
<th>Total # of detection opportunities</th>
</tr>
</thead>
<tbody>
<tr>
<td>Case 1</td>
<td>1</td>
<td>18</td>
<td>900</td>
<td>50</td>
</tr>
<tr>
<td>Case 2</td>
<td>2</td>
<td>9</td>
<td>450</td>
<td>100</td>
</tr>
<tr>
<td>Case 3</td>
<td>6</td>
<td>3</td>
<td>150</td>
<td>300</td>
</tr>
<tr>
<td>Case 4</td>
<td>18</td>
<td>1</td>
<td>50</td>
<td>900</td>
</tr>
</tbody>
</table>

*Table 1. TREX’13 processing parameters.*

Fig. 2 shows the resulting detection contacts for Case 1 and Case 4, which were formed from the spectrogram images through normalization, clustering and thresholding processes. Red and green contacts indicate positive and negative frequency shifts, respectively, in the heterodyne processing. Here the contacts are presented in terms of their time delays rather than frequency shifts. The opening target is clearly seen slanting from 1 to 6.5 second over the scenario. A few intermittent detection contacts for fixed features are shown vertically down the display. Case 1 produced 230 total detection contacts and Case 4 produced a total of 1307 contacts. Case 4 provided more detectability of the fixed features than did Case 1.

Next the data was input into a multisensor multi-target tracker [6] where good tracking performance was obtained with one clear E/R track and no false tracks. The tracker used an association gate of 99%, and process noise values ($q_x$ and $q_y$) of 0.05 m2/s3 in the NCV motion model. Fig. 3 shows a zoomed view of the track positional estimates obtained for Case 1 and Case 4, within a small geographical box. The red ellipses correspond to tracker “updates”, where detections are associated with the current track estimate. The green ellipses correspond to tracker “coasts”, where no detection was available to associate with the current track estimate and it has simply been projected in time and is expanded in size.

For Case 1, during this portion of the track, there is only one detection opportunity every other processed scan. This is because the E/R is only transmitting every other ping. Therefore, we see a “coast” between every update. Each coast produces an inflated AOU, as seen by the green ellipses. There are about three updates shown within this area. For
Case 4, the first observation to make is that there are many more measurements over the same time frame, consistent with the more frequent update rate (18 times as many observations as Case 1). We see that during the E/R’s off cycle, the consecutive set of coasts grow the size of the positional (green) AOU. Once detection updates are received, the AOU (red) compress quickly back down to smaller size. The overall sizes of these AOU are much smaller than seen in Case 1. AOU size for the cases shown here relatively small due to the large bandwidth signal used, however, when less bandwidth is available AOU sizes will be larger and the HDC processing will be better able to constrain their growth.

![Fig. 2. Detection contacts produced for Case 1 (left); detection contacts produced for Case 4 (right).](image)

![Fig. 3. View of sequential positional updates of the tracker for Case 1 (left) and Case 4 (right). Mean estimate (red markers) and AOU (red ellipses) for detection updates; mean estimate (green markers) and AOU (green ellipses) for coasts.](image)

The results of the AOU size (in square meters) for Case 1 and Case 4 can be compared in Fig. 4 (left). This is the AOU size ($p = 0.99$) calculated over a 5 minute portion of the scenario where the target track was maintained. We observe that the detection updates drive the positional AOU down, while tracker coasts drive the positional AOU up. Case 1
yield much larger AOU size than Case 4 when updating, by a factor of about 4-5 times. This is because the AOU growth has been contained by having additional detections with frequent update rate. From around 950 seconds, Case 1 shows detection updates on every cycle. This is because at this time during the run, the echo energy from the target is (more or less) equally split between two adjacent processing windows and detections are obtained. With a real target (rather than a simulated E/R target), there will be valid detection opportunities on every cycle, with similar probability of detection as recent cycles. The actual AOU will be more stable at the lower envelope of the data shown here. This data clearly shows the growth and inflation of the AOU without frequent measurement updates.

Fig. 4 (right) also shows the target speed uncertainty over the same 5 minutes period. For this run, the target is detected in the same beam, at the same bearing for the whole scenario so there are no changes in the target heading uncertainty. The target speed uncertainty can be obtained from the velocity uncertainty in a straightforward manner. We see the same effect; updates drive the uncertainty down while coasts drive it up. The more frequent measurements are better able to contain the speed uncertainty at a low level.

![Fig. 4. Positional AOU (left) and 1-sigma speed uncertainty (right) over 5 minutes segment. Case 1 coasts (cyan) and updates (magenta). Case 4 coasts (green) and updates (red). Speed uncertainty (1-sigma) over a 5 minute time period.](image)

### 5. SUMMARY

The potential advantage of HDC is an increased number of continuous detection opportunities, leading to improved target localization, tracking, and classification. With HDC sonar processing, the signal’s bandwidth is spread out over the entire transmission repeat cycle. Splitting the full band up into sub-bands by processing shorter time blocks, results in multiple detection opportunities per waveform cycle. Though the probability of detection and ranging accuracy may be lower for HDC sonar than for PAS (due to reductions in source level and processed bandwidth), a distinct advantage is that there is less time lapse between measurement scans. Such a rapid measurement update-rate effectively contains the growth of the target’s area of uncertainty (AOU) within a kinematic tracker. The processing and analysis of TREX’13 data confirms this hypothesis: that AOU growth is significantly constrained when HDS processing is employed. In the case of the TREX’13 data analyzed the data processed with fast update
rate (27 scans per minute) provided 4-5 times smaller positional uncertainty than the slow update rate (1.5 scans per minute). Decreases in velocity uncertainty were also shown. The impact of this analysis is that high data rate processing with HDC sonar may provide improved target localization, holding, maneuver detection. In addition, it reduces the possibility of track degradation due to the possibility of updating the track with false alarms. Larger AOUs will have increased probability of false alarms falling within them and being erroneously associated with the target track. However, if breaking the processing down in smaller intervals causes target echo SNRs to reduce sufficiently that they are no longer detectable, then the target localization and tracking benefits suggested here will not be realizable.

6. ACKNOWLEDGEMENTS

We acknowledge Dr. Paul Hines, Mr. Jeff Scrutton, and Mr. Stefan Murphy of Defence R&D Canada-Atlantic who designed and conducted the TREX’13 experiment; Dr. Dajun Tang, Dr. Todd Hefner, and Dr. Kevin Williams of the Applied Physics Laboratory of the University of Washington (APL-UW) who managed and led the TREX’13 trial, and Dr. John Preston of Pennsylvania State University’s Applied Research Lab who managed quality control and data collection for the experiment. We wish to acknowledge the officers and crew aboard CFAV QUEST and RV SHARP, and the APL-UW dive team for their support throughout the trial. Funding for this work was provided by ONR Code 32 and ONR Global-London.

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A Doppler Estimation Technique is Based on the Signals with Good Correlation Properties: Experimental Results

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Abstract: A Doppler estimation technique is presented. The technique is based on the application of the composite signal packet. Signal packet consists of signals with good correlation properties (e.g., Barker codes, M-sequence). The proposed approach provides a reducing in computational load and hardware cost in comparison with the method of ambiguity function. The experimental results confirming the efficiency of the technique are presented.

Keywords: Estimation of the Doppler effect, M-sequence-modulated carrier, autocorrelation.
1. INTRODUCTION

An integral part of research of sea areas has been the use of a variety of underwater autonomous vehicles (UAV). The only available means of communication with the UAV are acoustic waves. Significant influence of the Doppler effect on the propagation of acoustic signals in the waters complicates direct transposition of the radio-communication methods in underwater acoustics. Due to the low speed of the sound waves (~1500 m/s), the use of methods broadband modulation gives the Doppler shift by several orders greater than the radio. Mainly for impact estimation and compensation of the influence of the Doppler effect, researchers complicated structure of the signal packet, adding signals with «convenient» for estimating Doppler offset properties.

So, the signal packet, spreading from the source to the receiver, is subject to the influence of the Doppler effect caused by the movement of their source and/or movement of the receiver. In the frequency domain Doppler effect is the change in the carrier frequency signal:

$$\Delta f = f - f_0$$

(1)

where $f_0$ is the carrier frequency signal; $f$ - signal frequency is shifted by the Doppler effect; $\Delta = v/C$ - the value of which performs the function of the scaling factor; $v$ - relative velocity of motion of the source-receiver and $C$ - the wave propagation velocity. Estimation applied in underwater acoustics, traditionally, is based on the calculation of the mutual ambiguity function [1-4]. But this approach requires significant computing resources [5].

To reduce the computational load the evaluation by the authors of work [5] was proposed in the original algorithm, which is based on the detection of changes caused by the Doppler effect in the time domain and application tolerant Doppler signals. In the time domain, change the frequency of a signal equivalent to a change in the duration of the signal, the effect of «compression-tension»:

$$T_r = (1/(1+\Delta))T_s$$

(2)

where $T_s$ is the duration of the emitted signal; $T_r$ - duration of the signal after exposure to the Doppler effect. Thus, by measuring the duration of the signal at the receiver $T_r$ and to compare it with $T_s$, you can estimate the value of the scaling factor $\Delta$ and, further, to compensate for it. The group of authors [5] offer to use the signal packet, the following:

$s(t) = [x_{LFM}(t), x_{INF}(t), x_{LFM}(t)]$, here $x_{LFM}(t)$ Doppler-tolerant signal, linear frequency modulation signal (LFM, chirp), $x_{INF}(t)$ - information signal. Processing at the receiver is to calculate the «convolution» of the received packet $r(t)$ with $x_{LFM}(t)$:

$$R_{LFM}(\tau) = r(t) \otimes x_{LFM}(t)$$

(3)

Then we can calculate the time interval between peaks and is evaluated using (2). Further, for the compensation of the Doppler effect in a received signal packet is recalculated the sampling frequency of the received signal: $F_s' = (1+\Delta) F_s$, where $F_s$ is a sampling frequency of the receiver and using the linear interpolation in accordance with the new changes $F_s'$. We denote the operation of linear interpolation as

$$\tau_{PACKET}(t) = LI_\Delta[r_{PACKET}(t)]$$

2. DOPPLER ESTIMATION TECHNIQUE

The technique is based on the use of the signal packet, denote $s(t)$, consisting of $N = 2, 3, \ldots$ complex signals $x(t)$ with the «good» of the autocorrelation properties and application of the «convolution» received signal with itself (autocorrelation). Under the «good» of the autocorrelation properties is a form of the autocorrelation function, which has one of the
narrow peak at zero shifting along the time axis and minor amplitude sidelobes in other shifts. This technique was developed specially for the signals that are not tolerant to the Doppler effect, namely, for signals, carrier frequency, which phase-manipulation M-sequence (such signals is also called M-sequence on the carrier). But the principle of the technique can be used for other types of signals with the «good» of the autocorrelation properties, for example, the LFM-signals.

For a good example, consider the case for the M-sequences on the carrier. Let the signal packet consists of \( N = 2 \) complex signals \( x(t) \), then the structure of this signal packet can be written as follows: \( s_M(t) = [x_M(t), x_M(t)] \). If you calculate a «convolution» packet with itself (autocorrelation) \( R_M(\tau) = s_M(t) \otimes s_M(t) \), it will be a two significant peaks, separated from each other by the time interval \( T_s \) equal to the duration of one complex signal \( x(t) \). Thus, by measuring the duration of the signal at the receiver side, you can use the autocorrelation of the received packet \( r_M(t) \):

\[
R_M(\tau) = r_M(t) \otimes r_M(t) .
\]

(4)

Compensation of the \( \Delta \) in a received packet is performed according to the same scenario, above considered in [5].

3. EXPERIMENT

The experiment was conducted on acousto-hydrophysical POI FEB RAS Cape Shults 17 August 2013. The receiver was installed in one meter from the bottom at a depth of 10 m and at a distance of 50 meters from the beach. As a source had a cylindrical piezoelectric transducer. The characteristics of the source allows to work with the broadband signals at the center frequency of 2 kHz. The source was dipped with yachts which moved along a given trajectory during the entire experiment. Geographical coordinates of a trajectory of movement was registered by the GPS system. To the point of emission kept on one level, the source was equipped with an underwater wing. The measurements were carried out within 2 hours. On Fig. 1 presents the scheme of the experiment. Yacht moved along the trajectory of the triangle (removal of the receiver in parallel with the receiver and towards the receiver). Sound speed profiles (SSP), is represented in Fig. 1 (right), were measured at the points indicated green points in Fig. 1 (left), at the beginning and at the end of full-scale testing.

For sounding was used signal packet, which consisted of a block of complex signals (duration \( \sim 2 \) s), 1 second pause and tone at the carrier frequency 2 kHz (2 s) \( x_{\text{TONE}}(t) \):

\[
s_{\text{PACKET}}(t) = [s(t), 0, x_{\text{TONE}}(t)] .
\]

The block of complex signals consisted of: 2 chirp signals

![Fig. 1. (left) the Geographical layout of the experiment. (right) Profiles the speed of sound.](image-url)
$x_{LFM}(t)$ (center frequency of 2 kHz band from 1.6 to 2.4 kHz, the duration of a single chirp signal 0.2 s) located at the beginning and end of the block, and 2 M-sequences on the carrier $x_M(t)$ (the sequence length is 255 symbols, $f_0=2$ kHz, one symbol sequences represent about 4 period of the carrier frequency, duration, one M-sequences on carrier 0.51 s) located in the center of the unit: $s(t)=[x_{LFM}(t), x_M(t), x_M(t), x_{LFM}(t)]$. On the receiving side, the signal from the hydrophone was sampled on the sampling rate $F_s=48$ kHz and was recorded on a PC. Before treatment, all signaling packets are filtered in the frequency range from 1.5 to 2.5 kHz.

Processing tone in all packets was to calculate the spectrum of the received signal with the help of Fourier transformation and search for the frequency corresponding to the maximum of the spectrum. Processing unit of complex signals and subsequent evaluation was carried out under (3) and (4). In order to avoid confusion when comparing estimates of the same physical quantity (Doppler translation) obtained with the help of GPS measurements and several acoustic methods, the results of the assessments corresponding to different approaches were identified as shown in Table I. For transformation relative velocity $v_{GPS}$ in the Doppler shift of the carrier frequency was used the following values: $f_0=2$ kHz and the mean value of the velocity of sound in a waveguide $C=1513.87$ m/s. Using data from the GPS system were calculated the relative velocity of motion receiver-source $v_{GPS}$.

**Table I. The comparison of measurement results.**

<table>
<thead>
<tr>
<th>Measured physical value $f$</th>
<th>Designation of the estimates $f$ obtained with different approaches</th>
<th>Short description techniques</th>
</tr>
</thead>
<tbody>
<tr>
<td>$f = (1+\Delta)f_0$</td>
<td>$f_GPS = (1+\Delta_GPS)f_0 = \left(1+\frac{v_{GPS}}{C}\right)f_0$</td>
<td>- measurement of the displacement of the carrier frequency by using values $v_{GPS}$ obtained by the GPS system.</td>
</tr>
<tr>
<td></td>
<td>$f_T = (1+\Delta_T)f_0$</td>
<td>- measurement of the carrier frequency offset by using the estimates $\Delta_T$ obtained by the formula (4)</td>
</tr>
<tr>
<td></td>
<td>$f_TONE$</td>
<td>- measurement of the carrier frequency offset using the tone signal</td>
</tr>
<tr>
<td></td>
<td>$f_LFM = (1+\Delta_LFM)f_0$</td>
<td>- measurement of the carrier frequency offset by using the estimates $\Delta_{LFM}$ obtained by the formula (4).</td>
</tr>
</tbody>
</table>

**4. EXPERIMENTAL RESULTS**

On Fig. 2 presents the results of measurements of the Doppler shift of the frequency $f$, using a signal packet. Graphical comparison $f\_GPS$ with $f_T$, $f\_LFM$ and $f\_TONE$ shown in Fig. 2a. As the graphs estimation of the Doppler shift of the frequency, obtained by different approaches, give similar values. The lack of individual values of $f_T$, $f\_LFM$ and $f\_TONE$ is associated with relatively high levels of noise generated by moving several tourist boats in the waters. A rough estimate of the signal-to-noise ratio (SNR) is shown in Fig. 2c Comparing the accuracy of estimates between acoustic methods relative measurements of GPS and is shown in Fig. 2b.

For the results obtained by tone signal and proposed in the report of the technique can be concluded that the effectiveness of accuracy (effectiveness) estimates decreases when value of SNR is lower approximately by ~5 dB. The calculation is more precise thresholds can be done at a manageable level and the nature of the noise.

The results of the comparison accuracy of acoustic methods has shown that methods give similar to each other estimates (standard deviation $\langle|f\_GPS - f_T|\rangle=0.270$ Hz, $\langle|f\_GPS - f\_LFM|\rangle=$
0.304 Hz $|f_{GPS} - f_{TONE}| = 0.273$ Hz.; the percentage ratio of the number unfit for acoustic treatment packets to the total number of accepted packet $N[f_f] = 20\%$, $N[f_{LFM}] = 30\%$, $N[f_{Tone}] = 16\%$.

Fig. 2 The results of processing the signal packet and measurements in the GPS system. The comparison of the results of measurement $f$: a) acoustic methods $f_f$, $f_{LFM}$, $f_{TONE}$ and GPS data $f_{GPS}$; b) Evaluation of measurement accuracy of the acoustic data relative to GPS $|f_{GPS} - f_f|$, $|f_{GPS} - f_{LFM}|$ and $|f_{GPS} - f_{TONE}|$. c) signal-to-noise ratio $s(t)$ and $x_{TONE}(t)$.

To check on the validity of the estimates obtained using presented in this report methods has implemented the additional processing of audio data, which consisted in the following stages. At the first stage, was recalculated the sampling rate $F'_s$ in accordance with the received values $\Delta_f$, and using linear interpolation was converted adopted packet $\overline{r}_{PACKET}(t) = LI_s[\overline{r}_{PACKET}(t)]$. At the second stage, was calculated convolution:

$$\overline{R}_M(\tau) = \overline{r}_{PACKET}(t) \otimes x_M(t).$$

Also, was calculated convolution «pure» (without compensation Doppler effect) signal packet $r_{PACKET}(t)$ with $x_M(t)$:

$$R_M(\tau) = r_{PACKET}(t) \otimes x_M(t).$$

The results of the earlier steps, shown in fig. 3c and fig. 3d. From Fig. 3d shows that the evaluation of the Doppler shift of $\Delta_f$ the obtained using described in this report technique is correct as the compensation of the influence of the Doppler effect in accordance with $\Delta_f$, allows use of the properties of M-sequences on the bearing to measure the pulse characteristics of the channel.

Using the value of the frequency resolution M-sequences on the carrier $\Delta f_s = 1/T_s$ and formula $\Delta f_s = (v/C)f_0$, you can estimate what the value of the relative velocity signal is not correlated with the transmitted. In our case the threshold is $v = (\Delta f_s/f_0)C =$ (1.9608/2000) 1513.87=1.4842 in m/s (2.8875 knot). On Fig. 3 presents in detail the moment the «destruction» of the correlation of the received signal with emitted. Bold lines in Fig. 3c and d allocated transition moment to no correlation (this moment corresponds to the 195th packet). On Fig. 3a shows the change in the relative velocity, 195th packet matches the change from 1.48 to 1.5 m/s,
which agrees with the above estimates. On Fig. 3b presents the values of the maximal impulse
response \( m_M(i) = \max \left[ R_M(i) \right] \), here \( i \) is the sequence number of the received packet.

![Fig. 3](image)

Fig. 3 The results of experimental verification of the limits (thresholds) correlation received and
transmitted signal: a) change of the relative velocity of the source-receiver; b) change the
maximum value hotel implementations of the received signal; c) change the impulse response of
the channel without compensation Doppler effect; d) change the impulse response of the channel
with compensation of the Doppler effect.

5. CONCLUSIONS

To summarize, we can conclude that the results of the research conducted in conditions of
shallow sea in the presence of high noise level and pulse interference showed an assessment of
the Doppler shift of the frequency using the signal package, consisting of a signal with the
"good" of the autocorrelation properties. The value of the maximum relative velocity amounted
to 2.2 m/s (4.28 knot). A comparison with other acoustic approaches proved the efficiency,
correctness and effectiveness proposed in the report technique. This technique, similar to [5], it
allows to reduce the computational load when processing of acoustic data on the receiving side.
Moreover, permits not only Doppler-tolerant signals (chirp), and significantly expands the range
of complex signals suitable for estimating Doppler shift, for example, signals, generated on the
basis of the frequency and phase modulation.

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ONTOLOGY DESIGN FOR COOPERATIVE UNDERWATER TARGET TRACKING

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Abstract: Using information science terminology, cooperative target tracking for multistatic low frequency active sonar systems can be seen as inherited from a more general ontology for a distributed goal driven observation system. The term "distributed" refers to both, the spatial distance between components of the observation system and to local decision making onboard the platforms involved in the cooperative target tracking operation. The goal that drives the cooperative target tracking platforms can be formulated for the non-cooperative target case as a minimization of the number of behavioral choices left to the target. This formulation can be used as the main criterion for the design of the ontology. The collection of operationally relevant instances of the ontology can be used to construct the concept of operations for the tracking platforms. We will utilize this terminology to describe the findings from the analysis of multistatic simulations and sea trials.

Keywords: Low Frequency Active Sonar, Multistatic Sonar, Cooperative Target Tracking
INTRODUCTION AND SUMMARY OF PREVIOUS WORK

Low frequency active sonar enables the detection of submarines at large distances. However, modern submarines have a stealth design. Trained commanders are utilizing this stealth design in clever path planning to avoid detections as much as possible. For geometrical reasons the spatial distribution of a low number of acoustic low frequency sources leads to a decreased amount of degrees-of-freedom available to the submarine commander for his short-term and long-term plans. Surveillance systems are using automatic tracking and data fusion to extract detections of submarines from the stream of echoes generated by the active sound transmission. Within such automatic tracking and data fusion architectures, information theoretic metrics (e.g. information entropy [1]) can be used to calculate the information added by using multiple sources and receivers. Automatic target tracking and data fusion has been applied to data from scientific multistatic experiments and a performance gain has been demonstrated in terms of short latency (due to effective Doppler processing), high precision (due to triangulation), and less false alarms (due to target motion model based filtering) [2].

High-power acoustic sources are needed for long range detections. They are large and heavy and have a high demand on energy for a persistent operation. For logistical and maintenance reasons they probably have to have components with surface expression, at least temporarily. To avoid a large Doppler spread in echo data, acoustic sources should go at a rather lower speed (i.e. deployed for drifting or moored). A reasonable design for the acoustic sources is being deployed as stand-off sources, i.e. far away from the area where targets are expected and also far away from bistatic receivers. The target area is in the field of view of these receiving platforms, which can be designed to be small and of low weight. Persistence, even as unmanned vehicles without surface expression, could then be implemented through a proper logistic of assets exchange and refuelling. These covert and low cost receivers are of huge interest when a persistent and scalable solution for large area surveillance has to be implemented. Robustness of automatic estimation processes can be generated via multihypothesis data fusion and tracking algorithm [3][4]. A sketch of this scenario is given in Fig. 1.

In this paper, the development of a concept of operations is outlined which is adaptive to environmental changes (and changes in sensor performance accordingly). The sensor network consists of distributed mobile sensors with limited inter-platform communication. With regard to communication, we assume that the stand-off sources can be used to continuously distribute a small amount of coordination information via the acoustic underwater channel. The receivers should be kept as much as possible covert (in order to deny the stealth target to adapt to the source-receiver geometry), but having the capability to transmit over the low-frequency acoustic channel information on specific confirmed tracks. This information can be encapsulated into a few hundred bytes every minute. For this scenario, we have developed a coordination scheme and an outline how to evaluate its parametric settings [5]. Depending on the various parameter settings the scenario just described can be designed as a fair game in which both players (the target and the surveillance team) have equal chances to reach their goals. The operational goal for the target is to reach the left side of the surveillance area undetected. This goal drives the internal decisions on the choice of its behavioural model to avoid being trapped in the near future. The more alternative choices the target has the more difficult it is for the surveillance to trap the target. Hence, the surveillance team’s internal goal is to minimize the behavioural choices left to the target.
To actually play (at sea or in simulations) the game outlined in the previous section, a detailed description of the scenario is necessary. Since the parameter space is large and since there are numerous interdependencies and interconnectivities between the various equipment parts and between them and the environment, a tool set developed in Information Science is used to capture all details properly and independently of their relevance in a specific situation. This tool set is related to ontologies. The W3C Semantic Sensor Network Incubator Group [7] had two main objectives: “The first was to develop an ontology to describe sensors and sensor networks for use in sensor network and sensor web applications. The second was to study and recommend methods for using the ontology to semantically enable applications developed according to available standards such as the Open Geospatial Consortium's (OGCTM) Sensor Web Enablement (SWE) [8] standards.” One of these standards is for example the Sensor Model Language (SensorML) with the focus “to provide a robust and semantically-tied means of defining processes and processing components associated with the measurement and post-measurement transformation of observations. This includes sensors and actuators as well as computational processes applied pre- and post-measurement. The main objective is to enable interoperability, first at the syntactic level and later at the semantic level (by using ontologies and semantic mediation), so that sensors and processes can be better understood by machines, utilized automatically in complex workflows, and easily shared between intelligent sensor web nodes.” [9] Cooperative Underwater Target Tracking needs interoperability as an enabling necessity and intelligent sensor web nodes in order to
perform its task efficiently. Therefore, the findings that have led to the formulation of the SSN ontology are inherently valid also for Cooperative Underwater Target Tracking.

The SSN Incubator Group further describes the SSN ontology to be an answer to “the need for a domain-independent and end-to-end model for sensing applications by merging sensor-focused (e.g. SensorML), observation-focused (e.g. Observation & Measurement) and system-focused views. It covers the sub-domains which are sensor-specific such as the sensing principles and capabilities and can be used to define how a sensor will perform in a particular context to help characterize the quality of sensed data or to better task sensors in unpredictable environments.” The objective of an efficient Cooperative Target Tracking system is to task sensors in unpredictable environments, in the sense that not all environmental settings can be preplanned and stored for an automatic database requests driven execution of efficient Cooperative Target Tracking. The parameter space in such an implementation would be exorbitantly high. Therefore, in the Cooperative Target Tracking application a policy based reaction on newly sensed environments has to be implemented.

The experience from experiments at sea is that all details of the measurement setup have to be taken into account in terms of maintenance and risk analysis. The SSN ontology with its proposed amendments is capable to capture all details. In fact, in the maritime domain detailed information is already available for a lot of measurement systems [10]. However, although all information is available for a bottom-up design approach of a Cooperative Target Tracking system, reasoning mechanisms have to be designed and implemented which allow decisions (at design-time and at run-time) based only on the important aspects within the detailed description. Importance is meant relatively to the application goal and meant to be changing with the system states and with the environmental status during the execution. Reasoning is based on the ability to generate and evaluate physically realistic plans for future actions.

COMBINING ONTOLOGIES AND INFLUENCE DIAGRAMS

A top-down approach for the system design for Cooperative Underwater Target Tracking is helping to identify the importance of details. Engineering best practice, such as the multi-agent paradigm, can be used as a guideline through the top-down design process. In fact, an agent represents separately designed functionalities with explicit interdependencies to other agents. The agents’ concepts of operations describe these interdependencies from the perspective of the each agent, i.e. based only on the knowledge available to the specific agent. To understand how to avoid these interdependencies...
becoming critical for the overall effectiveness of the system, fair games are formulated and analyzed exactly for parameters at their critical value describing a phase transition of the associated functionality. This analysis enables, by extrapolation into non-critical domains, to predict the effect each functionality has in the entire system. In domains non-critical for one functionality, however, another functionality might become critical. If the design objective includes a gaming aspect (as it is, for example, the case for Cooperative Target Tracking for Smart Adversaries or as “Nature” can play, for example, a detection game against observation processes by adding noise to signals), the inherent interdependency between the players cannot be avoided, but has to be tackled by an appropriate methodology which we will discuss in the section on “Trading Methodology”. The prerequisite before the methodology can be applied is to clearly identify the gaming aspect formulated via the wish of the players (agents) to become independent. Having separately designed functionalities can be seen as a prerequisite for an agent to have a chance to achieve independence.

As developed in [5], the concept of operations extracted from the wish to become independent is further described in Fig. 1 (2nd paragraph) and summarized by the following three steps: STEP I: Creation of a clutter database, STEP II: Source deployment and scheduling, STEP III: Execution of the synchronized looks on clutter patches by covert receivers.

**TRADING METHODOLOGY**

In this section, we introduce the Trading Methodology as a tool how to construct efficient systems with a given effectiveness. The effectiveness of the game (described in the first section) has been analysed at the fair game setting of winning or losing with a 50%
chance. Of course, a Cooperative Target Tracking system should reach a higher rate of effectiveness. From the generated “CONOPS_50/50” we can easily add more layers of receivers, working independently from the first layer. The overall system reaches then a performance of (for example) 75%, e.g. calculated by assuming complete independence between two layers which is optimistic, but should be assumed here for the sake of simplicity of the discussion.

The resulting multi-layer is not very efficient, because the layers could operate in a coordinated manner (instead of working independently), meaning that by adding a coordination method less assets would be needed. This would generate a new CONOPS, the “CONOPS_75/25”. For generating this new CONOPS_75/25, the effectiveness is preserved, but the efficiency is increased by inserting coordination between two layers, making them become dependent. Therefore the name “Trading”: It is trading of dependence against efficiency under the constraint of a preserved effectiveness. Trading can be visualized e.g. in Influence Diagrams, where parts of the tree are connected in order to achieve higher efficiency. This redesign of the Influence Diagram is repeated to identify a platform (or sensor) which can disappear or which can be replaced by an available cheaper platform (or sensor). The execution of the Trading Methodology should be prepared in two ways: (i) the ontologies of all participating elements should include a cumulative efficiency measure (cost) such that recalculation of newly connected ontologies leads to a fast examination of the costs, (ii) a factored graph organisation of functionalities generated in separated design processes as we have already discussed in Fig 3b.

The effectiveness is preserved if the Trading Methodology steps do not change the criticality of the separately designed functionalities. I.e., all connections can be changed for the sake of efficiency as long as those critical for the original game are left unchanged, or in case critical connections are changed, the effect has to be compensated elsewhere in the Influence Diagram. For the latter, a fast prediction method for the effectiveness of systems with candidate changes is needed to check whether the compensation is achieved. The reason why fair games are analysed is that in a fair game with critical behaviour an analysis at this critical points can be used to extrapolate the findings made at this critical point by analytical mathematics without the need for further simulations or execution (which would be time consuming and/or expensive). This extrapolation can be used as the fast prediction method needed to look for possibilities to compensate for a predicted loss in effectiveness by changing critical connections.

**CONOPS FOR ENVIRONMENTALLY ADAPTIVE COOPERATIVE TARGET TRACKING**

Persistent maritime operations have to incorporate adaptability to the high variability (in space and time) of the meteorological and oceanographic (METOC) conditions at sea. This adaptability can be implemented via a two-step procedure: first a METOC forecast and then an adequate reaction of the target tracking system to these forecast results. The adequate reaction has to map the uncertainty associated to the forecast to the uncertainty of achieving the operational goal. Certainly, in a cooperative target tracking system, the adequate reaction of the system affects the coordination scheme between the assets participating in the cooperative target tracking. The coordination scheme describes the coordination architecture (including the number of participating assets) and the coordination method. Following the discussion in the previous section, we propose again the application of the Trading Methodology, starting with independent layers guaranteeing...
the necessary effectiveness also under changing environmental conditions, and then optimizing the coordination scheme towards efficiency.

As described in Fig. 1, we extend the original scenario to include changes in the receiver performance due to higher noise levels at their positions, for example caused by rain. The sonar equation combines in logarithmic units (i.e., units of decibels relative to the standard reference of energy flux density of rms pressure of 1 μPa integrated over a period of one second) the following terms: \( SE = (S - TL1 - TL2) - (NL - AG) + TS \) which define signal excess (SE) where: 
- \( S \): source energy flux density at a range of 1m from the source;
- \( TL \): propagation loss for the range separating the source and the target (TL1) and the target and the receiver (TL2);
- \( NL \): noise energy flux density at the receiving array;
- \( AG \): array gain that provides a quantitative measure of the coherence of the signal of interest with respect to the coherence of the noise across the receiving array;
- \( TS \): target strength whose value strongly depends on the aspect of the target to the source receiver pair.

Since the probability of detection (PD) is linked to SE, we elaborate on the example scenario by assuming an increase in NL such that a PD=0.5 is only given in a 3rd of the original operations area. This is described in Fig. 1. We further assume that the probability to forecast a rain front sufficiently early for a successful adaptation of the coordination scheme PF=0.75, i.e., with a probability of 0.25 a rain front is not forecasted sufficiently early for a successful adaptation of the coordination scheme.

From these assumptions, we want to generate an overall fair game, which could be further analysed and, as described in the previous section on the Trading Methodology, the analysis results could be further exploited according to new effectiveness and efficiency needs. Hence, in order to generate again a 50% overall chance of detecting the target in the environmentally adaptive scenario, the probability of target detection in case of correctly forecasted weather conditions has to be 75%. In the previous section, we inferred that this probability of detection can be assured by two independent layers. However, due to the higher noise level at all receivers, a probability of detection of these two independent layers is only 50% if each layer consists itself of three independent layers. In summary, six independent layers are needed. Six layers of independent surveillance would result in 18 sources. A closer look at the activities of the sources can examine that they transmit orthogonal replicas for matched filtering at the receiving units. Hence, three sources could transmit (without loss of effectiveness) the necessary source signals to drive (coordinate and activate special surveillance patches) the 12 AUVs. This application of the Trading Methodology is depicted in Fig. 4.

In summary, we can extract the environmentally adaptive concept of operations for the proposed Cooperative Target Tracking system: STEP I: Forecast capability for weather conditions to timely activate (sent, deploy, wake-up) more AUVs, STEP II: Creation of a clutter database, STEP III: Source deployment and scheduling, STEP IV: Synchronization.
of looks at clutter patches by covert receivers. STEP II to STEP IV are similar as developed in [5], but now they are extended to include to adapt to environmental changes.

CONCLUSIONS

The separation of design processes of functionalities in complex reactive systems leads to modularisation. The formulation of fair games between critical modules of a system design leads to the formulation of internal goals used for decision making at design time (likely to be centralized decision making) and at run time (likely to be distributed decision making) within the complex reactive system. As an example for complex reactive systems, a Cooperative Target Tracking system is analysed in this paper. The goal for the surveillance units (that leads to a fair game) is to minimize the number of behavioural choices left to the target. Taking this methodological design paradigm and applying it to operational relevant real scenarios can help to generate a concept of operations for the elements comprising the complex reactive system. As an example we elaborate on an environmentally adaptive deployment process which is part of the Semantic Sensor Network ontology and apply the methodological design paradigm. We show that with the methodology it is possible to connect Measurements of Performance with Measurements of Effectiveness even under the conditions of environmental uncertainty. This uncertainty is explicitly incorporated in a proper concept of operations, nevertheless a predictive and situation specific evaluation of effectiveness of the constructed sensor network is still possible. The mathematics in this paper are kept simple to demonstrate the steps of the developed methodology. The plan for future work is to expand the application of the methodology to more difficult examples and to use the methodology for the design of Computer Aided Reasoning helping operators to handle multi-platform tasks as well as for establishing automatic reasoning which enables autonomously cooperative operations.

REFERENCES

Session 21

Soundscapes and Measuring Noise

Organizers: Jennifer Miksis-Olds, Mark Prior and Kevin Heaney
APPLYING THE DYNAMIC SOUNDSCAPE TO ESTIMATES OF SIGNAL DETECTION

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Abstract: The field of underwater acoustics is currently struggling with how to define and apply the concept of a soundscape, which originated for use in the terrestrial environment, to the underwater environment. Sound (especially low frequency sound) travels greater distances underwater compared to in air, so sources from 1000s of kilometers away have the potential to significantly contribute to local soundscapes. Understanding the source contributions within a local soundscape can therefore be complex. In applying our understanding of the soundscape to signal detection, sound level trends, or noise impacts, it is first necessary to parameterize the acoustic environment. This work examines the impact of the dynamic soundscape and selected sound level parameters on estimates of signal detection area, also referred to as active acoustic space when considering effective acoustic communication between vocalizing animals. The range of signal detection was investigated at three site locations of the Comprehensive Nuclear Test-Ban Treaty Organization International Monitoring System. Transmission loss to each hydrophone was computed using the OASIS Peregrine parabolic equation model for a source within the upper 300m of the water column to be consistent with the location of vocalizing baleen whales. Daily, monthly, and seasonal soundscape measurements were incorporated into the sonar equation to estimate the variability in signal detection area as a function of sound level and time.

Keywords: Soundscape, signal detection, active acoustic space, noise level
INTRODUCTION

The study and application of the term “soundscape” has gained in popularity over the past five years, and soundscape ecology and soundscape orientation have been presented as new fields of research [1, 2]. The soundscape concept has been used to try to capture the dynamic spatio-spectral-temporal aspects of an acoustic habitat related to contributions from biotic, abiotic, and anthropogenic sources. Understanding the complex cumulative and synergistic contributions underlying the variations in underwater soundscapes is a necessary first step in relating changes in the environmental sound levels to signal detection and ultimately to effective communication and masking of marine animals using sound.

Studies are using aspects of the soundscape to estimate active acoustic space for vocalizing cetaceans, acknowledging the complexity in the environmental and biological parameters contributing to those estimates [3, 4]. Janik (2000) [3] used single values from previously published noise level (NL) data under two different sea states, whereas Clark et al. (2009) [4] used a single parameter measurement of the 5th percentile for NL over time to estimate active acoustic space. This work aims to demonstrate how much the signal detection area (DA) of a receiver varies as a function of soundscape over time. Here we consider only the physical contributions to calculations of signal detection areas (NL, transmission loss (TL), source level (SL)) without accounting for the biological characteristics of hearing needed to extend the DA estimates to estimates of active acoustic space for communication. We have examined variability of the soundscape and its impact on DA at three time scales (seasonal, monthly, and daily) and over three noise level parameters (daily minimum, median, and maximum NL).

METHODS

Signal detection areas (DA) around Comprehensive Nuclear Test Ban Treaty Organization International Monitoring System (CTBTO IMS) monitoring stations at Diego Garcia (H08: Indian Ocean), Ascension Island (H10: Atlantic Ocean), and Wake Island (H11: Pacific Ocean) were estimated using the passive sonar equation to determine the range along four bearings at which Signal Excess (SE) equaled zero (Equation 1). A constant source level (SL) of 180 dB re 1 μPa was used to be reflective of the range of estimated blue and fin whale vocalization source levels [4, 5, 6]. A constant detection threshold (DT) of 3 dB was applied, and directivity index (DI) and processing gain (PG) was assumed to be zero.

\[ SE = SL - TL - NL - DT + DI + PG \]  

Transmission loss (TL) for each season at each location was modelled along four bearings (0°, 90°, 180°, and 270°) using the OASIS Peregrine parabolic equation (PE) model for a receiver in the sound channel and a source within the upper 300 m of the water column to be consistent with the depth of vocalizing baleen whales (Fig.1). Peregrine is based on Michael Collins' split-step Padé PE marcher [7] (RAM), a widely used acoustic model for low to mid frequency undersea sound propagation modelling. Starting from Collins' RAMGEO 1.5 Fortran code, Peregrine has been ported to C, refactored for performance on modern computers, optimized for fully range-dependent problems, and is able to interpolate directly from geographically defined ocean field and bathymetry inputs. Sound speed profiles were obtained from The World Ocean Atlas. It includes an optional 3D azimuthal coupling operator, integrated time-domain output, range and
depth antialiasing, volume attenuation, and two-parameter sediment specification (thickness and grain size) among other improvements. For broadband, Nx2D, and 3D problems, Peregrine will automatically use all available CPUs in parallel.

![Image](image_url)

**Fig.1:** A) Receiver modelling TL output from the Peregrine PE model at CTBTO location at H10 N1 in the Atlantic Ocean at Ascension Island during the summer season. TL is shown as a function of depth and range. TL receiver output from the Peregrine PE model as a function of range around HA08 N1 (B) and HA08 S1 (C) in the Indian Ocean at Diego Garcia.

Noise level (NL) was calculated from acoustic recordings from a single north and south hydrophone at each CTBTO IMS monitoring location (see [8, 9] for details on CTBTO IMS monitoring stations and recording characteristics). NL measurements were made over three targeted 20 Hz bands (10-30 Hz, 40-60 Hz, 85-105 Hz) and are reported as spectral levels in decibels (dB re 1μPa²/Hz). Mean spectral levels were calculated using a 15,000 point DFT Hann window and no overlap to produce sequential 1-min power spectrum estimates over the duration of the dataset. Daily minimum, median, and maximum NL were identified from the 1440 minute averages calculated each day, which created a time series of daily values for each NL parameter.

Signal detection areas were estimated at three temporal scales: seasonal, monthly, and daily over 30 days. Seasonal estimates reflect estimates from 2011. Monthly calculations were made over a two year period from 2010-2011, and daily estimates were made for November 2011. Daily NL values were averaged over the appropriate time scale in three NL parameter categories: minimum, median, and maximum. These statistic parameters were selected to examine variability within the most extreme (min and max) and most likely encountered (median) DA estimates. Detection range estimates were calculated from the maximum range along each bearing where SE >0 in each noise category. Straight lines were used to connect the range points along the four bearings to form a polygon, and the area within the polygon was calculated from the bearing range lengths. The DA_max is reflective of the DA estimate based on minimum daily NL time series, whereas the DA_min reflects the minimum DA estimate based on maximum daily NL time series. DA_median represents the median DA estimate based on median daily NL time series. The % Difference was calculated for DA within each NL category (minimum NL, median NL, and maximum NL) and across seasons using the local category DA maximum (e.g. DA_max_minNL, DA_max_medianNL, DA_max_maxNL) as a standard of comparison and represents the percent difference between the maximum and minimum detection ranges within a NL category or across seasons by which to assess the percentage DA reduction based on increasing environmental sound levels within each NL parameter (Equation 2).
RESULTS

The seasonal noise levels had a greater variation within each frequency band and across seasons for maximum NL (2-12 dB re 1μPa²/Hz) compared to minimum NL (0-5 dB re 1μPa²/Hz) (Fig. 2). This corresponded to a 0-98% difference in DA across all seasons, frequencies, and locations for DAmax (Fig. 3) and a 0-100% difference for Damin. The seasonal NL (Fig. 2) was reflective of the monthly and daily NL averages across frequency and location. As this effort focused on the NL variability and how it translates into variation in DA, the remaining results present dB differences in NL as opposed to absolute levels. The dB difference was defined as the difference between the average maximum and average minimum values within each NL category time series. Averages reflect either monthly or daily averages depending on the time scale of analysis.

Average monthly NL dB differences across locations over the two year period of 2010-2011 were not significantly different between the monthly minimum and median NL for any frequency bands (df=10, paired t= -0.4- -2.0, p = 0.1-0.7) and ranged from 2- 8 dB (Fig. 4). The dB difference of the monthly maximum NL ranged from 7-30 dB and was significantly greater than the variation of minimum and median levels for all frequency bands at all locations (n=18, F(2)= 13-15, p ≤ 0.001). Monthly variation in DA over the course of two years varied from 0-98% depending on frequency and location (Fig 5, Table 1A). For all frequencies and locations, a portion of the distribution of DAmax and DAMedian overlapped. There was a varying amount of distribution overlap between the DAMin, DAMedian, and DAMax. Due to length constraints, the distribution of DA for 80 Hz is shown as an example (Fig. 5), and all frequencies for DAMedian are shown in Table 1A.

In November 2011, the dB difference did not differ for daily minimum and median NL for any frequency bands (df=10, paired t= -0.4- -1.5, p = 0.2-0.7) at each location and ranged from 1-12 dB. The dB difference in NL in the daily maximum NL ranged from 1-25 dB and was significantly greater than the daily minimum and median levels for all frequency bands at all locations (n=18, F(2)= 11-31, p ≤ 0.001). Daily variation in DA over the course of 30 days in November 2011 varied from 0-100% depending on frequency and location (Fig 6, Table 1B). For all frequencies and locations, a portion of the distributions of DAmx and DAMedian overlapped. It was uncommon to observe any overlap between the DAMin and DAMedian. Due to length constraints, the distribution of DA for 20 Hz is shown as an example (Fig. 6), and all frequencies for DAMedian are shown in Table 1B.

4. CONCLUSIONS

This work illustrates the order of magnitude differences in DA as a result of changes in the soundscape over time. This study did not address difference in DA as a result of transient sources such as passing vessels, rather the difference in DA observed here reflect changes in the ambient conditions averaged over a day. The percent difference in DA was a function of frequency and NL parameter selected. The greatest seasonal impact was observed at location H11 at Wake Island in the Pacific Ocean. Distributions of DA for the minimum and median NL parameters overlapped a majority of the time and can be considered comparable. The degree in variation for
Fig. 2: Minimum (A) and maximum (B) seasonal NL. Numbers above each frequency band indicate the dB difference across seasons for that frequency band and location.
Fig. 3: Estimates of $DA_{\text{max}}$ for the 5 modelled frequencies. Numbers above each frequency band indicate the % difference across seasons for that frequency band and location.

Fig. 4: Average monthly dB differences over two years from 2010-2011 at each location.

Fig. 5: Frequency distribution of monthly estimated DA for an 80 Hz signal over two years: 2010-2011. The % values in the legend represent the DA % difference within the designated NL category ($DA_{\text{max}}$, $DA_{\text{median}}$, $DA_{\text{min}}$).
Fig. 6: Frequency distribution of daily estimated DA for a 20 Hz signal over 30 days in November 2011. The % values in the legend represent the DA % difference within the designated NL category (DA_{max}, DA_{median}, DA_{min}).

DA computed with the maximum daily NL statistics was significantly larger than the variation computed with minimum and median NL values. The TL and resulting DA estimates from H10 N1 in the Atlantic Ocean were anomalous due to the bathymetry blockage at this location. In order to translate the physical estimates of detection area into communication space and masking impacts for vocalizing marine animals, the hearing capabilities related to frequency bands, thresholds, and integration time would need to be combined with the physical attributes examined here [4]. Based on the results of this exercise, it is clear that both humans and animals must constantly adjust their perceived range of signal detection to accurately interpret source location.

5. ACKNOWLEDGEMENTS

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Table 1. DA\textsubscript{median} statistics for all frequencies and locations over a (A) two year period 2010-2011 and (B) 30 day period in November 2011.

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SHIP SOUND MAPPING IN THE NORTH SEA

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Abstract: There is an increasing concern that anthropogenic underwater noise may have a negative impact on marine life. Governmental authorities are introducing regulations to address this problem. The European Commission, for instance, has adopted the Marine Strategy Framework Directive requiring EU Member States to achieve or maintain Good Environmental Status, regarding underwater noise amongst other forms of pollution. A task group (the technical sub-group on underwater noise) has formulated indicators for underwater noise pollution, resulting in an advice to monitor low frequency sound in particular frequency bands (sound pressure level in the third-octave bands centred at 63 Hz and 125 Hz). Model generated sound maps were identified as a monitoring tool to complement measurements.

Recently, the SONIC (Suppression Of underwater Noise Induced by Cavitation) project was awarded within the European Seventh Framework Program to develop tools to investigate and mitigate the effects of underwater noise generated by shipping activities. In this paper, we will present the SONIC approach to generate shipping sound maps. The sound map generation tool uses Automatic Identification System (AIS) as well as biological distribution data to generate maps representative of the sound exposure that marine mammals and fish would experience. This tool uses a fast acoustic model developed specifically for this purpose that was compared to normal modes and parabolic equation models. Results of the models compared with reference models are presented in this paper.

Keywords: Sound map, shipping noise, MSFD
1. THE SONIC PROJECT

There is a concern that the well-being of marine species can be influenced by background noise. The European Union Marine Strategy Framework Directive [1] requires Member States to demonstrate that levels of anthropogenic underwater sound do not adversely affect marine ecosystems. It is believed that propeller cavitation sound from ship traffic is largely responsible for low frequency ambient noise. A need has therefore arisen for an improved understanding of the correlation between ship design and operational parameters and background noise in the seas [3].

The aim of the SONIC (Suppression Of underwater Noise Induced by Cavitation) project is to develop tools to investigate and mitigate the effects of underwater noise generated by shipping, both in terms of the impact of an individual ship and of the spatial distribution of sound in a certain area from a large number of ships contributing to the sound field (a sound map). The main focus is put on understanding both cavitation and machinery related noise.

The project will investigate different techniques to model cavitation noise numerically and experimentally in dedicated hydrodynamic facilities, and techniques to measure noise levels in the sea. Finally, a methodology will be developed to calculate the contribution to the noise field of individual ships, as well as that of an ensemble of ships, i.e. a sound map for a given sea area.

The consortium of the SONIC project is led by MARIN (Maritime Research Institute Netherlands) and consists of:

- CNR-INSEAN (Consiglio Nazionale delle Ricerche, Istituto Nazionale per Studi ed Esperienze di Architettura Navale)
- HSVA (Hamburgische Schiffbau-Versuchsanstalt)
- Navantia
- Rolls Royce
- TNO
- University of Southampton
- University of Newcastle
- Wärtsilä
- ARTTIC

The SONIC project has furthermore been cooperating with a similar EC FP7 project (The AQUO project [2]) on comparing the output of sound mapping models. During a noise modelling workshop organised jointly between the two projects, benchmark scenarios for shipping sound map generation were defined.

In this paper, we present the shipping sound mapping approach adopted in SONIC. In the first section we detail our concept of a sound map. The second section describes the tool used for the generation of the sound maps and the fourth section deals with the bioacoustics components of the sound maps.

2. SHIPPING SOUND MAP CONCEPT

The term “sound map” suggests a notion of a geographical representation of a scalar parameter related to sound, which leaves room for much interpretation. To remediate this,
we propose a definition for a class of sound maps. To this end, we recall the purpose of the sound map, in the context of this study. The requirement for a sound map (Table 1) in our context is to give to a non-acoustic specialist a quantitative, global, geographical and temporal overview of the amount of sound due to a specific source (shipping), that in principle could be compared to threshold levels above which a risk exists of a specified effect on a specified marine species.

<table>
<thead>
<tr>
<th>Requirement</th>
<th>Specification</th>
</tr>
</thead>
<tbody>
<tr>
<td>Global</td>
<td>Basin size (North Sea)</td>
</tr>
<tr>
<td>Geographical</td>
<td>Gridded, Function of latitude and longitude</td>
</tr>
<tr>
<td>Temporal</td>
<td>At a given instant (snapshot), or averaged over a month, season or year</td>
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Table 1 Requirements and parameterisation of a sound map.

A common measure of the amount of the received “stationary” underwater sound radiated by ships is the mean squared sound pressure, defined as the average over a period of time of the squared instantaneous pressure, evaluated in a specified frequency band. It is often represented in a logarithmic form, expressed as the sound pressure level (SPL) in dB re $\mu$ Pa$^2$. Considering that the SPL distribution of underwater sound due to shipping is a five-dimensional quantity, (latitude, longitude, depth, frequency band and time), any two-dimensional geographical representation requires some averaging (or integration) in time, depth and frequency. Three options for averaging over depth and frequency are identified:

1. An absolute representation of total shipping noise is found via averaging over the whole water column and integrating over all frequencies bands.
2. Since different marine species are found at different depths and are sensitive to sound in different ways, a sound map as described in item 1 would not be representative of the amount of sound perceived by different marine species. Section 4 describes the generation of a sound map expressed in sound exposure level (SEL) while taking into account a species or group of species depth distribution and acoustic sensitivity curves.
3. The requirements given by the Marine Strategic Framework Directive [1] to member states focus on SPLs in the 63 Hz and 125 Hz third-octaves bands (defined as “a frequency band whose width is one tenth of a decade”). Generating a map using the SPL in this band is therefore of direct interest to policy makers.

Finally, the averaging time for the computation of these sound maps depends on their purpose. The evaluation of a specific ship-related activity at sea (for instance dredging) may require an integration of the order of hours of days, while the evaluation of freight related shipping may require an integration time in the order of weeks, months or years.

Because of the global aspect of a sound map it is impractical to obtain this from direct acoustic measurements, apart for assimilation or validation purposes. This seems to point towards the use of models for generating sound maps. Obtaining ship positions on a global scale is possible through analysis of data collected with the Automatic Identification System (AIS). A source level model in combination with source distribution and an acoustic propagation model is then used to compute SPL over the area of interest. In such a fashion, a snapshot of instantaneous positions of ships allows the computation of a sound map at a given instant. For generating a long term sound maps (i.e integrated over weeks
or months), we use ship density maps computed from averaged AIS data rather than individual ship positions.

3. SOUND MAPPING TOOL

A sound mapping framework has been developed for generating underwater sound maps and footprints due to shipping. While the framework has been set up for shipping sound, it is developed such that it can be easily extended to model other anthropogenic underwater sound sources such as piling, air-guns, explosives and natural sources such as wind and rain. The framework is coupled to various external databases to define the environment on a global scale. Relevant environmental parameters are the bathymetry, sound speed profile, surface loss as a function of wind speed, sediment properties (density and attenuation) and volume attenuation as a function of temperature, acidity, salinity and depth.

Ships are modelled as points sources at a defined depth below the sea surface. The shipping source level is currently estimated using the model proposed by Wales and Heitmeyer [4]. A more accurate shipping source level model using information available from AIS data is under development in the SONIC project. For the spatial distribution of the sources, both average shipping density maps and AIS snapshots can be used as input. The density map models the average number of ships of a certain class within a grid cell for a specified time interval. This makes it possible to approximate the temporally averaged SPL in a computationally efficient way. The advantage of using snapshots is that is allows studying the temporal variability and statistics of the sound. The disadvantage of introducing the temporal variability is the increased computational effort. The computational effort can be reduced by pre-computing the propagation loss (PL) in a lookup table. However, in order to keep the data size of the PL lookup table within bounds, compromises are required in the number of dimensions. The preferred modelling approach is therefore dependent on the application.

Besides the coupling of the framework to external environmental databases, various acoustic propagation models are included. This makes it possible to easily compare different PL models and validate the modelling approach. It also allows the use of different models for different frequencies, optimizing both accuracy and computation time. Models included in the sound mapping framework are KrakenC [5], RAM [6] and SOPRANO [7]. SOPRANO is an hybrid propagation algorithm based on mode and flux theories for the shallow waters. It shows good agreement with normal mode theory. SOPRANO takes into account range dependent water-depth and sediment type for iso-velocity cases. It calculates incoherent PL including the depth dependent properties of wave theory. Figure 1 illustrates the validation of the depth average broadband SPL computed with SOPRANO (left), against the parabolic equation model RAM (right) on a synthetic scenario generated around the Skagerrak area and featuring water depths down to 500 m. The computation time for generating the sound maps for SOPRANO was in the order of tens of minutes, while the computation time for the RAM model was in the order of days. The 3D distribution of the SPL is computed by means of interpolation from 2D slices.
In the final step of the modelling process, sound maps are generated by reducing the number of dimensions of the SPL from four (latitude, longitude, depth and frequency) to two (latitude, longitude), by means of weighting the mean squared sound pressure with the vertical animal distribution and hearing sensitivity. This weighting is discussed in more detail in section 4.

The modular character of the sound mapping tool allows both fast computation of sound maps for a wide range of frequencies/large scale modelling, while maintaining the flexibility to study more complex/computationally expensive scenarios. Also, the flexible character of the model makes it straightforward to develop and test new modelling concepts and study the effect of uncertainties in the input, propagation and post processing on the sound map.

4. IMPACT ON MARINE LIFE

The mean square sound pressure computed by the propagation model over a given period of time are four dimensional quantities (Latitude, Longitude, Depth, Frequency). To provide the end user with a synthetic representation reflecting the quantity of sound perceived by marine life, we choose to average the mean squared sound pressure over the frequency and depth dimensions using weighting functions characteristic of the density and sensitivity of the concerned animals. These weighting functions were generated separately for marine mammals and fishes.

- Marine mammals:

Depth distributions of animals were predicted using the Vertical Movement (Dive) Module of the SAFESIMM programme [9][10]. SAFESIMM (Statistical Algorithms for Estimating the Sonar Influence on Marine Megafauna) is a simulation-based framework for modelling the numbers of animals affected by underwater sounds such as those used in geological and geophysical exploration. It was originally developed for assessing the impacts of sonar on marine mammals but has been expanded to integrate with other sound sources including air guns and pile driving. The software tool uses the latest research on
the effects of sound on marine mammals together with data on the distribution, abundance and hearing characteristics of these species. The core code for SAFESIMM is written in the statistical programming environment R.

The Vertical Movement module within SAFESIMM was utilised to predict the proportion of time animals would spend in different regions of the water column over a range of bathymetry values. The vertical dive movements are based on parameters from an internal database, which was populated from an extensive literature review that includes all the marine mammal species that occur in the North Sea. For example, specific parameters determine the dive depths, dive durations, surface durations and swimming speeds of each species.

The marine mammal species likely to occur in the North Sea were determined using the SAFESIMM species database.

In the selection of frequency weightings for the marine mammal species, a range of established approaches and the latest threshold weightings were considered. These included those developed by Southall et al. [11], Finneran and Jenkins [12] and the recent NOAA draft guidelines [13].

- Fishes:

  From a marine biological point of view the southern North Sea is considered a continental shelf sea or neritic zone. The zone is considered stable and well-mixed. Grouping of fishes to assess noise impact can therefore not be based on depth ranges, but should start from ecological grouping of fish species. For a shallow shelf sea it is sensible to distinguish three main categories: pelagic, benthopelagic, and benthic species. Pelagic species are often (not always) shoaling. For most species their depth range is not limited to the upper layers. For the North Sea it is therefore safest to assume that these species use the whole water column, but avoid the bottom (5 m above the bottom). Benthopelagic species live in the vicinity of the sea floor, but have neutral buoyancy. Given the shallowness of the North Sea, they can be found in the whole water column. Benthic species are strictly bottom-dwelling and have negative buoyancy. They will rarely be found outside the 0-5 m depth range (from the bottom).

  These three ecological groups will be further subdivided in hearing groups (Table 2). In the pelagic zone herring-like fishes (Clupeiformes) have best hearing, with high sensitivity and hearing in the higher frequencies (30 Hz-3000 Hz). Other pelagic species such as mackerel are probably slightly less sensitive and have a narrower frequency range (100 Hz-2000 Hz). Among benthopelagic species cod-like species (Gadiformes) are known to have a high hearing sensitivity in a low frequency band (30 Hz-400 Hz). Other benthopelagic species, such as salmon and sea trout have a medium sensitivity, but a broader frequency range (30 Hz-1000 Hz). Benthic species often lack a swim bladder and can therefore hear only the particle displacement component of sound. Flatfish (Pleuronectiformes) typically have a low hearing sensitivity in the range 20 Hz-200 Hz. Benthic species that do have a swim bladder (gurnards, gobies) generally have low or medium sensitivity and a broader hearing range (100 Hz-1000 Hz).
Table 2. Tentative grouping of North Sea fish for sound impact assessment. S. = (hearing) sensitivity, H = high, M = medium, L = low; F. = frequency

5. CONCLUSION

In this paper, we have presented an approach to shipping sound mapping that includes different types of representations can provide policy makers with a synthetic overview of the quantity of sound either in the bands of interest or emphasising the effect of sound on both marine mammals and fishes. This is achieved by weighting modelled sound pressure levels over depth and frequency range of species of interest. For marine mammals, we rely on statistical modelling of dive profiles using the SAFESIMM model, along with frequency weighting curves adopted from the literature. A new, tentative hearing and depth grouping is proposed for fish species present in the North Sea.

The future tasks in the SONIC project include the generation of maps for specific actual scenarios. Sound maps will also be generated using altered AIS data to quantify the influence of mitigation measures, such as shifting shipping lanes and altering ship operation, on the resulting sound maps.

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PRACTICAL SPREADING LAWS: THE SNAKES AND LADDERS OF SHALLOW WATER ACOUSTICS

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Abstract: Geometrical spreading laws are widely used in underwater acoustics because they provide - if chosen carefully - an accuracy that is sufficient for many applications (source characterisation, impact assessment, sound mapping, regulation) for negligible computation time. The simplest and most widely used form is that corresponding to spherical spreading, with propagation loss, PL(R) = 20log_{10}R dB, which can provide a suitable approximation for deep water. In shallow water, propagation is influenced by multiple reflections from the seabed and sea surface, and a modification is then needed. The resulting effect depends on the kind of source (e.g., continuous or transient) and its directivity (e.g., monopole or dipole). The result often simplifies to the form PL(R) = (A + Blog_{10}R) dB, where the values of A and B depend on the conditions. If the source level is known, the received SPL can then be calculated for a continuous source using sound pressure level, SPL(R) = SL - PL(R) = SL - (A + Blog_{10}R) dB, but the expected behaviour depends on source directivity, so the values of A and B need to be adjusted accordingly. For a transient source, there are no simple expressions for sound pressure level, but the sound exposure level can be related in a similar way to the energy source level. Guidance is provided for the choice of A and B in shallow water for different activities, including seismic surveys, shipping, explosions, pile driving and use of active sonar. Guidance is also offered on where not to use these simple rules, such as for the calculation of SPL for short transient sources and the pitfalls associated with applying far-field concepts such as source level to distributed sources such as pile drivers.

Keywords: Shallow water propagation, Spherical spreading, Cylindrical spreading, Mode stripping, Monopole source, Dipole source, Airgun, Pile driving, Ship, Explosion
1. INTRODUCTION

Modelling the propagation of sound in shallow water has various applications such as source characterisation, environmental impact assessment, sound mapping and regulation. Given some property of a source such as its source level (SL) in a certain frequency band, what is the sound pressure level (SPL) at some receiver at position \( x \) relative to the source? This question can be answered if propagation loss (PL) is known because the three quantities are related via

\[
\text{SPL}(x) = \text{SL} - \text{PL}(x)
\]  

(1)

Propagation loss is a transfer function defined as the difference between SL and SPL. For a monopole source in free space, this transfer function can be written in the form

\[
\text{PL} = 10\log_{10}\left(\frac{W}{4\pi I(s)/r_0^2}\right) \text{ dB},
\]

where \( W \) is the sound power radiated by the source, \( I(s) \) is the intensity at distance \( s \) from the source and \( r_0 = 1 \text{ m} \). For the special case of spherical spreading from a monopole source, the intensity \( I(s) = W/(4\pi s^2) \), and PL simplifies in this situation to

\[
10\log_{10}(s/r_0)^2 \text{ dB}.
\]

Using the upper case symbol \( R \) to denote the dimensionless ratio \( s/r_0 \), this can also be written

\[
\text{PL} = 20\log_{10} R \text{ dB}.
\]

Sometimes one encounters ‘\( \text{PL} = N\log_{10} R \text{ dB} \)’, often with \( N = 10, 15 \) or 20. For \( N = 20 \) this expression for PL corresponds to spherical spreading formula, but what does it mean for values of \( N \) other than 20?

Some progress can be made towards answering this question with a dimensional argument. The physical quantity represented by any level in decibels is the argument of the \( 10\log_{10} \) operation before dividing by the reference value. In the case of the transfer function, PL, the physical quantity being expressed in decibels is the reciprocal of the propagation factor [1], which has dimensions of length squared and is equal to the ratio of an area (that area into which the sound has spread at the point it reaches the receiver) to a solid angle (that solid angle from which the sound originates). The reference value of this physical quantity is \( r_0^2 = 1 \text{ m}^2 \). Considering the expression

\[
\text{PL} = (A + B\log_{10} R) \text{ dB},
\]

the above dimensional argument has implications for the relationship between \( A \) and \( B \). Basically, if \( A \) is the level of some physical quantity \( a \), the value of \( B \) determines the dimensions of \( a \). If \( B = 20 \), \( a \) is dimensionless. More generally, using the notation \([L]\) to indicate a physical quantity with dimensions of length, the dimensions of \( a \) are \([L]^{2-B/10}\). Examples are cylindrical spreading, for which \( B = 10 \) and \( A = 10\log_{10}([L]/r_0) \), and mode stripping, for which \( B = 15 \) and \( A = 10\log_{10}([L]/r_0)^{1/2} \). In both cases the distance represented by \([L]\) is proportional to the water depth. Section 2 describes basic properties of underwater sound for a continuous monopole source. Effects of proximity to the sea surface and of transients are described in Secs. 3 and 4, respectively. Distributed sources are considered in Sec. 5. A range independent environment is assumed throughout, with uniform sound speed and negligible absorption in the water. In reality, deviations from these assumptions will result in deviation from ‘practical’ geometrical spreading.

2. BASIC PROPERTIES OF UNDERWATER SOUND

The quantities SPL, SL and PL are introduced in turn in this section. SPL is defined in terms of the RMS sound pressure, \( p_{\text{RMS}} \), as [2]
The reference sound pressure is $p_0 = 1 \, \mu\text{Pa}$, the physical quantity expressed here as a level is mean square sound pressure $p_{\text{RMS}}^2$ and the corresponding reference value is $p_0^2$.

SL is a property of the source closely that is closely related to its radiated sound power $W$. For a monopole in free space, of uniform density $\rho$ and sound speed $c$, these two quantities are related via $\text{SL} = 10\log_{10}\left(\frac{\rho c}{4\pi} \frac{W}{(p_0^2 r_0^2)}\right) \text{ dB}$. More generally, SL can be defined in terms of the source factor $S$ [1]

$$\text{SL} = 10\log_{10} \frac{S}{p_0^2 r_0^2} \text{ dB}$$

where $S = p_{\text{FF}}(s)^2 s^2$, and $p_{\text{FF}}(s)$ is the RMS sound pressure at distance $s$ from the source and in its far field (i.e., unaffected by reflections from boundaries). The source factor is not a property of the sound field but of the source. The far-field sound pressure is that sound pressure that would exist in the far field of that source if it were placed in an unbounded loss-free medium with the same density and sound speed as the true medium at the source position, and excited with identical motion on all acoustically active surfaces as in the true medium.

In Eq. (4), the physical quantity expressed as a level is the source factor $S$. That quantity has dimensions (pressure times distance) squared, and its reference value is $p_0^2 r_0^2 = 1 \, \mu\text{Pa}^2 \text{m}^2$. Propagation loss is defined as [1] $\text{PL} = \text{SL} - \text{SPL}$. Equation (1) follows from this definition. Further, substituting Eqs. (3), (4) in Eq. (1), it follows that

$$\text{PL} = 10\log_{10} \frac{S}{p_0^2 r_0^2} \text{ dB}$$

At a small distance $s$ from the source, the area into which the sound has spread is $4\pi s^2$. The solid angle at the source ensonifying this area is $4\pi$, so the reciprocal of the propagation factor is $4\pi s^2 / 4\pi = s^2$, corresponding to spherical spreading ($B = 20; A = 0$):

$$\text{PL} = 10\log_{10} \frac{s^2}{r_0^2} \text{ dB}$$

Further away (several water depths) from the source the area becomes $2\pi rH$, where $r$ is the horizontal distance and $H$ is the water depth. For a seabed critical angle $\psi$, the vertical range of angles that propagates in the cylindrical spreading region is $-\psi$ to $+\psi$, corresponding to a vertical opening angle of $2\psi$. The range of azimuth angles is $2\pi$. The solid angle is therefore $2\psi$ multiplied by $2\pi$, i.e., $4\pi \psi$, leading to the expression for cylindrical spreading in shallow water at frequencies above the shallow water cut-off frequency: [1, 3]
\[ PL = 10 \log_{10} \frac{r}{r_0} \text{ dB} + 10 \log_{10} \frac{H/(2 \psi)}{r_0} \text{ dB}, \tag{7} \]

and corresponding to \( B = 10 \) and \( A = 10 \log_{10}[H/(2 \psi r_0)] \). In Eq. (7) and subsequent equations in this section, \( r \) is the horizontal range (not the slant range). The change in distance variable from \( s \) to \( r \), made for convenience, makes little material difference at the longer distances of relevance here.

After multiple bottom reflections the energy in steep paths close to \( \psi \) is dissipated more rapidly than paths close to horizontal, reducing the effective propagation angle to \[ \theta_{\text{eff}} = \left( \frac{\pi H}{4 \eta r_0} \right)^{1/2}. \] Replacing \( \psi \) with \( \theta_{\text{eff}} \) in Eq. (7) (the area is unchanged) gives the usual “15logR” mode stripping formula: \[ PL = 10 \log_{10} \frac{r^{3/2}}{r_0^{3/2}} \text{ dB} + 10 \log_{10} \left( \frac{\eta H / \pi}{r_0^{1/2}} \right) \text{ dB}, \tag{8} \]

corresponding to \( B = 15 \) and \( A = 5 \log_{10}[\eta H/(\pi r_0)] \), implying that \( A \neq 0 \) unless \( \eta H = \pi r_0 \).

If \( A \) is arbitrarily set to zero in Eq. (2), for a silt or sand seabed, the magnitude of the resulting error is typically up to about 5 dB in very shallow water \((H \sim 1 \text{ m})\), increasing to 15-20 dB for deeper water \((H \sim 100 \text{ m})\).

Figure 1 (left) shows \( PL(r) \) calculated using Equations (6) to (8). These equations may be used with Eq. (1) to predict SPL for a continuous source, such as a sonar projector, deployed sufficiently far from all boundaries that its radiation impedance is unaffected by the presence of any boundary. Effects of the sea surface and of transient sound are discussed in Secs. 3 and 4, respectively.

![Graph showing propagation loss vs range for monopole and dipole sources.](image)

**Fig.1**: Propagation loss vs range. left: monopole \((H = 30 \text{ m}, \psi = 0.5 \text{ rad}, \eta = 0.3 \text{ Np/rad})\); right: dipole (frequency = 300 Hz, \( z = 1 \text{ m}; D = 10 \text{ m}; \) other parameters as for monopole). Vertical cyan lines separate regimes of spherical spreading (left), cylindrical spreading (middle) and mode stripping (right).

### 3. PROXIMITY TO THE SEA SURFACE

The derivation in Sec. 2 assumes a monopole source not close to a boundary (in terms of acoustic wavelength), and especially not close to the sea surface. Departures from this assumption are considered next. The sound field in the vicinity of a point source in close proximity to the sea surface is strongly influenced by the reflected field, which
experiences a $\pi$ phase change, resulting in a Lloyd mirror interference pattern. In this situation, Eq. (1) still holds in the form $\text{SPL} = \text{MSL} - \text{PL}$, where MSL is the monopole source level [1]. The effect of the sea surface is to modify the received field for a given source level (because of interference between the direct and surface reflected paths) in such a way that Eqs. (6)-(8) no longer hold in general. In this situation the dependence of PL on range depends on frequency. The effect of this frequency dependence on the values of $A$ and $B$ is considered next.

For spherical spreading, it is useful to define a coordinate system with its origin at the centre of the dipole, on the sea surface directly above the monopole (at depth $z$). If $s$ is the distance from the origin in this co-ordinate system, and $\theta$ the elevation angle such that for receiver depth $D$, $\theta = \arcsin(D/s)$, the spherical spreading PL can be written [1]

$$\text{PL} = 10\log_{10} \frac{s^2}{r_0^2} \text{dB} - 10\log_{10} \left[4\sin^2\left(kz\sin\theta\right)\right]\text{dB}$$

(9)

where $k$ is the acoustic wavenumber in water. This can be written in the form of Eq. (2), with $B = 20$ and $A = -10\log_{10}[4\sin^2(kz\sin\theta)]$, although the value of $A$ is only a constant if the elevation angle $\theta$ is also a constant. If instead of keeping the angle fixed it is the depth $D$ that is kept fixed, in the long range (small argument) limit the values of the constants become $B \approx 40$ and $A \approx -10\log_{10}[4(kzD/r_0)^2]$.

In the cylindrical spreading region the applicable power law becomes [4]

$$\text{PL} = 10\log_{10} \frac{r}{r_0} \text{dB} + 10\log_{10} \frac{3H}{r_0} \left(\frac{k^2z^2r_0^3}{r^3}\right) \text{dB}$$

(10)

(assuming $kz < \pi/4$, here and for the remainder of Sec. 3), corresponding to $B = 10$ and $A = 10\log_{10}\left[3H / (k^2z^2r_0^3)\right]$). The corresponding equation in the mode stripping region is [4]

$$\text{PL} = 10\log_{10} \frac{r^{5/2}}{r_0^{5/2}} \text{dB} + 10\log_{10} \frac{4(kz)^2 (n^3 / \pi H)^{1/2}}{r_0^{4/2}} \text{dB},$$

(11)

corresponding to $B = 25$ and $A = 10\log_{10}\left[4(kz)^2 (n^3 r_0^3 / \pi H)^{1/2}\right]$. Equations (9) to (11) are plotted in the right hand graph of Fig. 1.

All of the above assumes that SL is the monopole source level, in other words it is the source level of a sound source in isolation, without the contribution of (e.g.) the surface image if the source is close to surface. Sometimes cited for such a source is the source level of the dipole formed by itself plus surface-reflected image combined. Such a dipole source level ($\text{SL}_{dp}$) is not suitable for use in Eq. (1). [5]

Current measurement standards for underwater radiated noise of surface ships do not take into account the actual propagation loss at the test location. The reported quantity is a so-called radiated noise level (RNL), which represents the measured SPL, scaled by the distance between ship and hydrophone under the assumption of spherical spreading (Eq. 6). [5] This RNL is not suitable for use in Eq. (1). Given sufficient information about the measurement geometry and environmental parameters, the propagation loss could be estimated for a monopole below the sea surface. This concept provides an acceptable
approximation for noise radiation that is due to propeller cavitation, but is less accurate for other source mechanisms, like machinery noise, which is radiated from the ship hull as a distributed source in the proximity of the water surface – see also Sec. 5.

4. TRANSIENT SOURCES

All equations presented up to this point apply to a steady state situation, implying that the duration of the sound is sufficiently large for the steady state to be reached. A different approach is needed for sounds that do not reach a steady state. In that situation it is more useful to cast equations in terms of time-integrated quantities such as sound exposure instead of time-averaged ones such as mean-square sound pressure. More specifically, the simplicity and functional form of Eq. (1) is retained for a transient, with the same expressions for PL(r), if SPL is replaced with sound exposure level (SEL) and SL with the energy source level (ESL). With these same substitutions in Fig. 1 it follows that those graphs (monopole or dipole version as appropriate, depending on acoustic frequency and proximity of source to the sea surface) apply also to transient sources such as explosions (if far enough away to avoid the non-linear shock wave) and airguns.

Seismic airgun sources typically comprise several tens of airguns in an array. Each airgun releases a bubble of compressed air when commanded to do so. If all the airguns are at the same depth they are usually fired simultaneously.

For the purposes of imaging, the airgun array is usually represented as an array of monopoles (one for each bubble) and each monopole is characterised by a source function called the “notional source signature”. The notional source signatures can be determined using near-field hydrophones [6] or they can be modelled [7, 8] or they can be approximated by notional source signatures measured for individual air-guns [9].

It is usual to characterise arrays of airguns by their time-domain source signature, defined as the far-field product of sound pressure and distance: $S_{dp}(t) = p(r,t) \cdot r$, a property that is independent of distance from the source, where the distance $r$ is usually taken to be in a direction vertically beneath the airgun array. This quantity is related to dipole source energy level (ESL$_{dp}$), by implication also in the vertical direction, by the formula

$$ESL_{dp} = 10 \log_{10} \int \frac{S_{dp}(t)^2 \cdot dt}{p_0^2 \cdot r_0^2 \cdot t_0^2} \text{dB} \quad (12)$$

As with RNL and SL$_{dp}$ mentioned previously, this quantity needs to be converted to a monopole source level before it can be used in Eq. (1).

5. DISTRIBUTED SOURCES

Up to this point the concepts of PL and SL are used to characterise the received field via SPL = SL – PL or SEL = ESL – PL. Both PL and SL are defined in terms of the far field of the source. In this section we consider sources that span the entire water column. Such distributed sources are less well understood than the sources considered previously. We have in mind pile-driving equipment typically used for offshore construction in shallow water. Because of the difficulty in identifying the far field of such sources, the concepts of SL and PL then need generalisation before they become applicable, with or
without simple spreading laws. For the case of impact pile driving, Reinhall and Dahl invoke a complex-phased array of point sources of fixed strength and linearly varying phase that depends on depth of the point source in order to predict the radiated sound field as function of range from the pile. The phasing establishes angular and depth dependence in the pressure field that is verified with field observation.

An important range scale emerges (Fig. 2a) from this approach equal to \( r^* = \text{depth}/\tan \theta \), where angle \( \theta \) is related to the ensuing Mach wave; a typical \( \theta \sim 17^\circ \) puts \( r^* \) at about 3 times the water depth. For ranges \( r/r^* < 1 \) the field strength depends strongly on measurement depth, \( D \), whereas for \( r/r^* > 1 \) this is less so, with significant implications on the allocation of measurement effort for environmental noise monitoring for bridge or wind farm construction. A depth average of the field from impact pile driving (Fig. 2b) is consistent with cylindrical (not spherical) spreading at least for ranges up to \( r^* \), as expected from energy-conservation. For a situation involving uniform water depth, consistent with the assumption of a range-independent environment made throughout, the reduction of SEL with range is expected to go as \( \text{SEL} = \text{constant} - (10 \log_{10} R \ dB + \alpha r) \), where the exponential decay arises from multiple reflections from the seabed. The constant in this expression is a logarithmic measure of the sound energy radiated as a consequence of the hammer impact. It is not source level and does not have the dimensions of source level.

\[
\text{SEL} = \text{constant} - (10 \log_{10} R \ dB + \alpha r)
\]

Fig. 2: a) Measurements (symbols) of SEL as function of scaled range \( r/r^* \) for two hydrophone depths compared with PE simulation (lines), where results in both cases are plotted as a function of range is scaled by \( r^* \). b) Depth-averaged measurements of SEL at three ranges (averaging depth 4.9-10.5 m) compared with equivalent averages of PE simulated data made at 1 m range intervals. The dashed line represents cylindrical spreading. For details, see Ref. [13].

6. CONCLUSIONS

The basic equations for propagation loss look (and are) simple: Eq. (1) and related equations for SPL, SEL; Eqs. (6) to (13) for PL, but there are some rules. If propagation loss is written in the form \( (A + B \log_{10} R) \ dB \), the values of \( A \) and \( B \) are not arbitrary and not independent of one another. Special cases considered include a surface ship, airgun and pile driving. In all cases except pile driving there is a region (“regime”) of spherical

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spreading followed by cylindrical spreading and then mode stripping. The “regime” determines the value of $B$, which in turn determines the functional form (dependence on frequency, water depth, etc) of $A$.

Possible numerical values for $B$ include 10, 15, 20, 25 and 40. The “constant” $A$ can be equal to 0 dB, but in most situations takes some other value (see especially Secs. 2 and 3). Pile driving is a special case that is less well understood than the other sources considered, and for which a generalisation of the concept of source level and propagation loss is the subject of ongoing research.

REFERENCES

SIGNAL GROUPING BY CORRELATION OF CEPSTRA

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Abstract: Data recorded on deep-sound-channel hydrophones, connected by cables to shore stations, are described. The data are gathered for purposes of nuclear test-ban monitoring and are routinely processed to determine noise spectra and the distribution of sound across arrival azimuth. These parameters can be used to help describe the underwater sound field at the hydrophone stations but a fuller depiction of the ocean soundscape requires some level of identification of the sources responsible for the various components of the sound field.

A method is described in which groups of similar discrete signals are identified via correlations between their cepstra. Cepstral correlation coefficients are used to form clusters of signals with clustering thresholds designed to identify the largest non-trivial groups. The method is applied to data gathered at a hydrophone station operated by the Comprehensive nuclear-Test-Ban Treaty Organisation (CTBTO) at Diego Garcia in the Indian Ocean. Large clusters are formed and these are identified as containing signals from baleen whales. Residual inter-cluster correlations are identified as resulting from the presence of overlapping signals of different types.

The views expressed are those of the authors and do not necessarily reflect the view of the CTBTO Preparatory Commission.

Keywords: Soundscapes, ambient noise, cabled observatories, CTBTO, seismic
1. INTRODUCTION

The comprehensive nuclear-test-ban treaty organisation (CTBTO) operates the International Monitoring System (IMS); a global network of seismic, infrasound, radionuclide and hydroacoustic sensors. The non-radionuclide data from this network are continuously processed and analysed to identify discrete signals that stand out from the background noise. Signal properties such as arrival time and direction are used to estimate the times and locations of the events that generated those signals. This is done initially using automatic signal processing algorithms and the results of these are subsequently refined and corrected by expert analysts. The outcome of this process is a daily bulletin that comprises a list of the events detected by the IMS [1]. The majority of events in these bulletins are earthquakes but also present are volcanic eruptions, meteor explosions (bolides) as well as anthropogenic explosions such as mining blasts and nuclear test explosions.

The hydroacoustic part of the IMS includes hydrophones deployed below sub-surface buoys in the SOFAR channel [2] and attached to shore stations by cables supplying power and data connection. Each cable has three hydrophones deployed in an equilateral triangle in the horizontal plane. This configuration is referred to as a ‘triad’ and most IMS hydrophone stations have two triads, one to the north of the island hosting the shore station and one to the south. This twin-triad configuration avoids blockage by the host island and insures combined 360-degree coverage for the station.

Hydrophones are monitored continuously and have been in operation for more than a decade in some cases [3]. Routine processing is carried out at CTBTO’s International Data Centre (IDC) in the frequency band between 3 Hz and 100 Hz. Signals detected against background noise on the hydrophones are generated by a variety of sources including earthquakes, in-water explosions, ice-breaking and marine life, especially baleen whales [4]. The automatic processing algorithms used by IDC must detect discrete signals and produce an initial classification of the type of source from which they originate. Seismological terminology is used for this purpose and signal type is described in terms of a ‘phase’ descriptor which for hydrophone data may be H, T, P or N. H-phase signals originate from in-water explosions (both natural and anthropogenic) and propagate to the hydrophone via waterborne paths. T-phase signals are generated by events in the Earth’s crust whose seismic vibrations couple into waterborne sound at coastal boundaries then travel through the water to the hydrophone. P-phase signals are also generated in the earth’s crust but travel to the hydrophone through the crust. The term ‘N-phase’ is used to denote ‘noise’ and, for the processing applied to IMS data, this broad category covers signals from marine mammals, ice-breaking events, air-gun surveys and other human sources.

To understand current – and predict future – performance of IMS hydrophone stations, it is necessary to have a good description of the acoustic field in the ocean. Previous attempts to describe this “ocean soundscape” [5] have used long-time-base spectrograms to describe the level of ‘diffuse’ background noise against which detections must be made. They have also used histograms of signal arrival time and azimuth to show how discrete noises are concentrated from particular directions. These descriptions often allow the nature of the generating sources to be identified. For example, signals concentrated around an azimuth pointing to an area of high earthquake activity are likely to be seismic in origin. Alternatively, signals that are grouped tightly in time but spread widely in azimuth suggest a nearby, moving source that may be related to shipping or marine mammals. This
information can be a useful addition to the automatic signal processing routines that produce the initial H, T, P or N-phase classification.

While these basic descriptors of the ocean soundscape are undoubtedly useful, it would be advantageous to produce a finer level of detail by which signals of similar types are automatically grouped. This paper describes an attempt to group signals on the basis of similarities in their cepstra. Some basic details of cepstral processing are first given then an example dataset is described. The clustering method is described and illustrated by application to the dataset. The merits and weaknesses of the approach are described and its potential future utility discussed.

2. CEPSTRAL PROCESSING

The real cepstrum [6] of a signal is produced by inverse Fourier transforming the logarithm of the absolute value of the Fourier transform of the signal. This process allows separation of multiple versions of the same signal, such as echoes or propagation multipaths. The technique is used in routine processing at CTBTO to identify the presence of bubble-pulses in signals from underwater explosions. These pulses are caused by the collapse of the gas bubble formed by the explosion and are often separated from the noise of the main detonation by times less than the duration of that noise. Thus, the signal generated contains two overlapping sub-signals. These can be represented mathematically by the convolution of a band-limited impulse with a Dirac comb containing two delta functions spaced by the bubble-pulse delay time. The Fourier transform of this convolved signal is given by the product of the Fourier transforms of the impulse and the comb. When logarithms are taken, this product becomes a sum and the signal and the comb are separated. The inverse Fourier transform of this logarithm contains a spike at the time separation of the main detonation and the bubble collapse. This delay can be used to characterise the explosion depth and yield [7].

The original application of the method [6] used the “power cepstrum” where the results of both forward and inverse Fourier transforms are modulus-squared. However, in the work reported here, the ‘real cepstrum’ was used in which the logarithm is taken of the absolute value of the Fourier transform and sign is conserved after the inverse Fourier transform. Thus, the cepstrum, \( c(q) \), of signal \( s(t) \) is defined here as

\[
 c(q) = \text{Re}\left[F^{-1}\left(\log|\text{abs}(F(s(t)))|\right)\right] 
\]  

(1)

Where \( F(x) \) denotes the Fourier transform of signal \( x \) and \( q \) is the quefrency variable that can be related to ‘time delay’ in processing applications associated with echo-extraction.

Cepstral processing is widely used in speech recognition where its sensitivity to the presence of echo structures allows it to identify features particular to an individual’s organs of articulation. It therefore represents a method that is associated with the identification of similar (but not identical) signals and which is routinely applied to IMS hydroacoustic data. This concurrence motivated the study reported here.
3. DATASET

To investigate the use of cepstral processing, a dataset was selected from the hydrophone data gathered at the northern triad of the Diego Garcia IMS hydroacoustic station. This station is in the Indian Ocean and shows periods of high incidence of marine mammal activity, particularly at the northern triad. The dataset covered the 20th April 2010 to the 5th May 2010 and was chosen to include a period in which whale sounds were frequently observed. CTBTO’s automatic processing algorithms identified 742 discrete signals during the thirteen-day period. The mean and median separations between signals were 1500 s and 465 s respectively, indicating that signals tended to arrive in bursts separated by quieter periods. The most intense period of activity was observed on the 22nd of April when 20 signals were recorded in a single hour.

Cepstra were calculated for all signals and the left-hand pane of Fig. 1 shows an image of these cepstra stacked into a single matrix giving cepstral amplitude as a function of quefrency and signal number. The colour scheme in the image uses blue for negative, orange for positive, passing through white for zero.

![Fig. 1: Stacked cepstra for all signals (left) and cross-signal cepstral correlation coefficient matrix (right).](image)

The cepstra show two main classes of behaviour. Signals such as the first 75 or so have little structure to their cepstra and maximum absolute values are small. On the other hand, many cepstra, such as those between indices 100 and 200, oscillate with quefrency between higher values and show significant similarity with other signals. Inspection of the waveforms and initial phase-identifications for these signals revealed that those with significant cepstral structure were marine mammal calls while those without were normally seismic signals.

The right-hand pane in Fig. 1 highlights the similarity between some signal cepstra. It is a shaded image of the matrix of correlation coefficients for all pairs of cepstra and the colour scale is set to cover the range between -0.1 (dark blue) and 1 (dark orange). The
signals with cepstral structure are shown to correlate well with each other and the clusters of indices in which they occur are marked by orange squares centred on the leading diagonal. These indicate signals which have high cross-correlation and may be considered similar. The correlation matrix also shows off-diagonal orange squares, indicating that there is considerable similarity between some signals with widely spaced indices. The orange squares, however, are not completely solid and contain blue lines where dissimilar signals occurred during the bursts of signals with high cross-correlation.

Fig. 2: Stacked cepstra for signals (left) and cross-signal cepstral correlation coefficient matrix (right). Signals re-ordered to cluster those with similar cepstra. Red lines delimit clusters

Fig. 2 shows the same data as Fig. 1 but with the signal order changed from a simple ‘soonest first’ approach to an ordering designed to group together signals with similar cepstra. This grouping was achieved by interpreting the correlation coefficient between each pair of signals as an inverse “data distance” so that signals with similar cepstra were treated as being close together in data space. The signals were then formed into agglomerative clusters using standard methods [8] and signal indices reordered on the basis of the number of the cluster to which they belonged. Clustering requires a user-specified threshold to be applied when assessing the significance of inter-signal data-distances. There is no definitive way that this can be set by consideration of signal properties. Instead, the threshold was varied from low to high and the number of resulting clusters was recorded. Very low thresholds resulted in each signal belonging to its own ‘cluster of one’ while very high thresholds resulted in all signals belonging to the same cluster. Between these extremes, the number of clusters changed with threshold, often in a step-like manner. The final threshold was set to be the value that resulted in the largest change in the number of clusters. This was interpreted as being the threshold that marked the most significant change in the correlation between signal cepstra.

The correlation matrix in Fig. 2 shows how signals grouped together to form two main groups. The sorted matrix is more concentrated about the leading diagonal than its
equivalent in Fig. 1 but it is notable that some off-diagonal correlations remain, so that the clusters are not completely separate.

Fig. 3 shows spectrograms of 200-s duration for signals associated with three groups of signals in the sorted dataset. In each case, the automatic detection is centred at the 100-second mark and the colour scale is logarithmic so that the dynamic range covers four orders of magnitude. All three signals contain lines of high energy concentrated at specific frequencies and the top two spectrograms are centred on these – the left-hand signal showing a line at 25 Hz and the right-hand signal showing a line at 14 Hz. The bottom-right signal is centred on a stack of lines and a similar but not identical stack is also present in the top-right signal. When sped up by a factor of 16, to make them audible, the lines sound like whistles and moans and the stacks sound like croaks. All three signals show some similar features and this goes some way to explaining the residual off-diagonal correlation observed in Fig. 2. Long-duration lines at around 102 Hz are present in all three signals, as are the pairs of lines at around 61 Hz and 36 Hz.

The orange squares in the sorted cepstral correlation matrix retain a chequer pattern, indicating that within the cluster there are some signals that show low correlation with many others. Inspection of these signals reveals them to be from reasonably quiet periods where a single sub-signal is present. For example, a signal that contains an isolated stack will correlate well with signals containing a similar croak. These signals, in turn, correlate well with signals containing the various line features. The original signal is consequently clustered with signals with which it has ‘second order’ connection, via the signals that contain both stacks and lines.

The data in Fig. 3 provide an illustration of the limitations of any method that attempts automatically to cluster signals into groups of similar types. Ocean noise often contains a superposition of signals from many sources and properties averaged over these mixtures will inevitably lead to residual correlations between clusters. However, the figure also demonstrates how much can be achieved and, for example, the lack of residual correlation
between the two largest clusters is a clear indication of the presence of two different classes of signal

Fig. 4: Cluster number plotted as a function of signal time order. Red bars above zero have heights proportional to the number of members of each cluster.

Fig. 4 shows dots indicating the cluster number and unsorted index for all signals detected in the dataset. The red bars above the zero line are a histogram of cluster number and the four main clusters are identified as being (in order of decreasing membership) numbers 144, 126, 186 and 32. The figure shows an interesting relationship between the two largest clusters in that they are initially present in the same timespan – that covered by signal indices 100-200. Fig. 2 shows that these two clusters have very little residual cepstral correlation and are the closest thing to orthogonal signals observed in the entire dataset yet they are temporally coincident. The clusters are, as shown in Fig. 3, associated with different types of marine mammal call but it is beyond the scope of the work reported here to comment on whether these are different calls from the same animals or calls from different animals. Clusters 186 and 32 are shown in Fig. 4 to be present at similar times and when the two largest clusters tend to be absent. Inspection of the signal types in clusters 186 and 32 shows that they contain seismic signals and periods of shipping noise. The signals are often seen against a background of marine mammal noise but the signal properties are likely to be dominated by the loudest part of the signal. Consequently the ‘either/or’ behaviour of the clusters is a simple consequence of the fact that in periods of loud seismic or shipping signals, whale noise no longer determines signal cepstral properties. This can be seen in Fig. 2 where signals associated with clusters 186 and 32 show little cepstral structure. It is interesting to note the change in observed clusters for signal indices in the regions 100-200 and 400-500. In the first of these periods, both clusters 144 and 126 are present but in the second, only 144 is observed.
4. DISCUSSION

The method described here, by which signals are grouped into clusters according to similarity in their cepstra, has been shown to be a useful way of identifying broad categories of signals. This is a potentially useful addition to the existing methods used to produce a broad signal categorisation in terms of the H, T, P and N-phase labels. An indication of this can be seen by the presence of multiple, reasonably distinct clusters of signals which might otherwise be grouped together as “whale noise”. However, this identification of signal clusters is far from complete. Overlapping signals and corruption by long-term background noise inevitably mean that signals are a mixture of contributions from different sources. This results in the residual inter-group correlations that are present even in the sorted data. Despite this, the approach of cepstral correlation clustering shows promise and should be considered as one of many potential methods by which signal grouping could be performed so as to clarify the ocean soundscape.

REFERENCES

PASSIVE ACOUSTIC MEASUREMENTS OF SNAPPING SHRIMP FROM A REEF MONITORING FEASIBILITY TEST IN ARUBA

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Abstract: In December 2013, TNO made underwater measurements in Aruba to assess the feasibility of reef health monitoring using passive acoustics; this work was conducted in collaboration with Aruba Ports Authority, Aruba Marine Park, and Aruba Reef Care Foundation. Ambient noise recordings were made at various locations around the reef over a period of several days and marine biologists performed a survey to assess the species present and to identify healthy and unhealthy regions of reef. This paper presents results from the ambient noise analysis. The underwater soundscape was found to be dominated by snapping shrimp noise. The sounds from other species known to be present were not observed, presumably due to their low levels and due to measurements made at a distance from the reef. We present an analysis of the shrimp noise, for which we observed a diurnal trend, and conclude with some comments on the feasibility of reef health monitoring.

Keywords: passive sonar, reef health monitoring, snapping shrimp

1. INTRODUCTION

Coral reefs are the natural habitat of a large bio-diversity of animals and vegetation. They also contribute to the economy of coastal areas by providing spectacular diving spots and constituting natural wave breakers. For these reasons, it is important to protect coral reefs from negative anthropogenic influences.

Passive acoustic reef health monitoring is a potential means of protecting these environments. The general idea is that, by measuring the underwater soundscape, it is possible to infer the presence, abundance, and behaviour of sound-producing species that
inhabit or rely upon the reef. It is hoped that the state of reef health can then be assessed and monitored using this information, providing quantitative evidence and justification for remediation. An essential step in this procedure is the separation of biological sounds of interest from other underwater sounds, including anthropogenic noise (e.g., from shipping) and environmental noise (e.g., wind, rain, etc.). Once this is achieved, the next challenge is to develop robust metrics that can be used to infer characteristics of the population of species present and the final challenge is to determine if and how these data can be related to the state of a reef’s health. The complete procedure is illustrated in Figure 1.

2. **SOUNDSCAPE MEASUREMENTS**

In December 2013, we took initial steps towards assessing the feasibility of reef monitoring using passive acoustics by making measurements of the underwater soundscape in Aruba. It should be noted that these experiments were opportunistic and coincided with a primary goal of experimentation on passive diver detection [1], and this imposed a number of limitations. It was desirable to sample the underwater soundscape thoroughly in space and time to allow spatial and temporal variations and trends to be observed. However, due to time constraints, it was necessary to compromise with several short-duration measurements at a variety of locations and a couple of long-duration measurements at a single location.

The underwater acoustic measurements were made in and around the port of Oranjestad, Aruba. Broadband signals up to 100 kHz were recorded using hydrophones and a data acquisition system from AguaTech [2] and SMID Technology [3]. Nine short-duration measurements of approximately 2-3 min each were taken between 11:00 and 15:00 on 10 December at the locations 1-9 indicated in Figure 2(a). For these measurements, the hydrophones were suspended roughly 6 m below a boat with its engine off. Two long-duration measurements were taken at the tip of the pier between locations 1 and 4: overnight from 10-11 Dec and over the weekend from 13-16 Dec. For these measurements, the hydrophones were fixed to the pier at roughly mid water depth.

The water depth in the region varies from 5-15m close to the port and in the shipping lane and >20m beyond the reef. The reef was inspected visually by divers and areas of healthy and unhealthy reef were identified; these areas are indicated in Figure 2(a) and
photographs of the reef are shown in Figure 2(b). The unhealthy and healthy areas are at depths of approximately 10 m and 10-20 m, respectively. The long-duration measurements and many of the short-duration measurements (1-5) were made near the port where the diver detection experiments were being conducted. However, some measurements (6-9) were made near the reef. Locations 6 and 7 were near the unhealthy area, location 8 was near the healthy area, and location 9 was between the two areas.

3. DATA ANALYSIS AND RESULTS

Spectrograms of the soundscape measurements are shown in Figure 3(a) and Figure 4(a). The loudest anthropogenic noise observed was due to (intermittent) acoustic emissions from the engines and active sonar systems of cruise ships. Some small boats (e.g., a water taxi) also contributed to the noise. Aside from anthropogenic noise, the soundscape was dominated by impulsive noise created by snapping shrimp [4]. The shrimp noise was present at all measurement locations and at all times. The sounds from other species known to be present, including parrot fish, grunts, and lobsters, were not observed. Presumably this is due to their much lower levels and the stand-off distance between the hydrophones and the reef. Future dedicated experiments will be necessary to observe the acoustic emissions from these species.

3.1. Snapping Shrimp Noise

We present an analysis of the snapping shrimp noise to illustrate parts of the methodology outlined in Figure 1. The aim is to observe spatial and temporal trends in the soundscape that can be associated to the population and behaviour of the species. Some comments on how this might relate to reef health are given at the end.
Metrics for quantifying impulsive bio-acoustic noise have been already proposed by other researchers, e.g., for investigating the sounds made by clams [5]. We use a similar metric here to quantify the snapping shrimp noise. The metric is computed as the median kurtosis over an interval of $M$ samples, i.e.,

$$K_{\text{med}}[n] = \text{median}\{K[n+i]\}$$

for $i \in [-M/2, M/2 - 1]$, where the sample kurtosis

$$K[n] = \left(\frac{1}{N} \sum_{i=-N/2}^{N/2-1} (d[n+i]-\bar{d}[n])^4\right)\left(\frac{1}{N} \sum_{i=-N/2}^{N/2-1} (d[n+i]-\bar{d}[n])^2\right)^2$$

is evaluated on a smaller window of length $N$ samples, $d[n]$ is the discrete-time signal, and

$$\bar{d}[n] = \frac{1}{N} \sum_{i=-N/2}^{N/2-1} d[n+i]$$

is the sample mean. This statistical measure quantifies the “peakedness” and tail length of the distribution. A kurtosis of 3 corresponds to a Gaussian distribution, whereas a higher value (leptokurtic) indicates that the noise is more impulsive. Thus, we assume that higher kurtosis values correspond to more shrimp activity.

A median interval of 5 s and a sample window length of 0.5 µs were selected for computing the kurtosis metric and the data was first high-pass filtered above 30 kHz to limit the influence of low-frequency anthropogenic and environmental noise.

**Spatial trends**

The distributions of kurtosis values are shown in Figure 3(b) for the short-duration measurements at locations 1 and 4-9. The results show that the kurtosis is higher away from the port (locations 6-9) and that there is an increase from the unhealthy reef area (locations 6 and 7) to the healthy area (locations 8 and 9). However, there is insufficient information to attribute these trends directly to the shrimp population density; there is clear interference from anthropogenic noise in some of the measurements (e.g., from active sonars) and the influence of varying environmental properties (e.g., water depth, seafloor properties, sound-speed profile, etc.) have not been accounted for. These results are, therefore, inconclusive and more controlled experimentation and analysis is necessary.

**Temporal trends**

For the long-duration measurements, an automatic classifier was implemented to remove the most dominant source of anthropogenic noise caused by cruise ships. A schedule of cruise ship activity was available and this was used to segment the data into periods with and without cruise ships present (see Figure 4(c)). Two signal features were computed over these periods: the signal power and the kurtosis metric described above. Figure 5 shows scatter plots of these features for the segmented data, and these suggest decision boundaries of signal power $< 40$ dB re $1 \mu$Pa and kurtosis $> 3$ for signals corresponding primarily to snapping shrimp activity. The interference from smaller vessels has been ignored and this should be considered in the future, possibly necessitating the use of additional classification features.

Compared to the spatial trends, the temporal trends are expected to be less heavily influenced by the environment since the seafloor properties are constant and the tidal
range is low (approx. 1 m). For this reason, here we attribute the observed temporal variations directly to the shrimp. However, we have ignored temperature variations and this assumption needs to be revisited in future work.

The features of the signals attributed to shrimp activity were computed and averaged over the cumulative period of three days. The resulting distributions of signal power and kurtosis are shown in Figure 6. Several daily trends can be observed: there is a relatively high level of activity during daylight hours, a peak of increased activity for approximately 1 hour during sunset, followed by a linear decrease in activity during the night to a minimum at approximately 1 hour before sunrise. The peak at sunset and the increase at sunrise is in agreement with the dusk and dawn choruses that have been observed elsewhere (e.g.,[6]). However, typically more activity is observed during the night than during the day, and our results are in contrast to this.

4. CONCLUSIONS AND DISCUSSION

Passive acoustic reef health monitoring relies on a capability to measure the acoustic emissions from sound-producing species, to isolate and quantify these signals, to relate them to population and behavioural characteristics, and finally (if possible), to use this information to infer a state of reef health. We have outlined this general procedure, illustrated parts of the methodology based on analysis of the snapping shrimp noise, and exposed some of the complications and challenges associated with this idea.
Due to the opportunistic nature of the experiments and domination of the soundscape by snapping shrimp, we were unable to observe the subtle acoustic emissions from other sound-producing species known to be present. Furthermore, we had insufficient measurements to observe any convincing spatial trends in the snapping shrimp noise. We did, however, observe some interesting temporal trends, which are at least partly supported by observations made elsewhere of a dawn and dusk chorus.

Analysis of the snapping shrimp noise is interesting for other applications (e.g., diver detection). However, the literature suggests that snapping shrimp are unreliable indicators of reef health and that fish are better indicators [6]. For this reason, future dedicated experimentation is necessary to document the acoustic emissions of relevant species and to investigate measuring these emissions effectively in the environment.
Figure 6 – Daily trends of the shrimp noise metrics at the tip of the pier between locations 1 and 4: (a) average power and (b) median kurtosis. The solid black line is the average, the coloured region indicates one standard deviation, and the dotted lines indicate two standard deviations. The day and night periods are indicated in (c).

REFERENCES

SOUND MAPS OF THE DUTCH NORTH SEA FOR NATURAL AND ANTHROPOGENIC SOUND SOURCES

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Abstract: Regulations for protecting and preserving the marine environment (e.g. in the USA and EU) often require investigating the potential effect of anthropogenic sound on marine life. The origin of underwater sound can be natural as well as anthropogenic. To assess the potential importance of various types of sounds, we constructed sound maps for the Dutch North Sea for both natural sources (i.e. wind and rain) and anthropogenic sources (shipping, explosions, and seismic surveys). Different sources affect different species, because of different frequency ranges or because of their distribution in time (e.g., continuous or intermittent; changing suddenly or gradually). Our maps take into account different averaging times, different swimming depths and frequency-weighting according to different hearing sensitivities. The underwater acoustic propagation is modeled mathematically by combining Weston’s average intensity method and adiabatic normal mode theory, which can provide fast and accurate results without calculation of normal mode eigenvalues and tracing rays. These maps, combined with information on species distribution and their physiological and behavioural reactions to sound, provide a useful indicator for understanding the impact of sound on marine life in the Dutch part of North Sea.

Keywords: sound maps, soundscapes
1. INTRODUCTION

Regulations in the USA [1] and the European Union (EU) [2] aim to protect and preserve the marine environment. The EU’s Marine Strategy Framework Directive (MSFD) requires member states (MS) to achieve or maintain Good Environmental Status (GES) by 2020. Specifically, the wording of Descriptor 11 requires “underwater noise to be at levels that do not adversely affect the marine environment”. The MSFD further requires MS to monitor “trends in the ambient noise within the 1/3 octave bands 63 and 125 Hz (centre frequency)”[2]. The impact of natural and anthropogenic sounds on marine life is a multidisciplinary concern which requires the collaboration of different disciplines such as ecologists, behavioural biologists and acousticians. The results of these investigations can provide clearer guidelines for the seismic surveys, controlled underwater explosions, pile driving and shipping activities, which are the major anthropogenic sound sources.

In this work, we investigate the contribution of each sound source to total sound field in the North Sea. A mathematical model is constructed for each sound source. Next, a fast analytical hybrid sound propagation model [3] is used in order to estimate the distribution of the sound in the Dutch part of North Sea. We consider variation in propagation loss related to frequency range and receiver depth, taking source depth, sediment absorption and bathymetry into account. In this way, sound maps are generated for different temporal scales (annually, seasonally or weekly averaged) and receiver depths. Hearing sensitivities and distribution data of different marine mammal and fish species groups determine the temporal and spatial resolution of these maps. When the sound maps are combined with these biological inputs, the final maps can provide insight into the potential for sound impact on marine life in the Dutch North Sea. These maps can also be useful to complement acoustic monitoring data and marine spatial planning [11].

2. SOUND SOURCES

Various natural sound sources (wind and rain) and anthropogenic sources (shipping, underwater explosions, seismic surveys) are considered. Each source model may require different mathematical approximations and can be described by different metrics. Source level (SL) or energy source level (SL_E) can be used to describe the sound source characteristics. These are the source factor (S) and energy source factor (SE) respectively, expressed as levels in decibels

\[ SL = 10 \log_{10} \frac{S}{p_{ref}^2 t_{ref}^2} \]  
\[ SL_E = 10 \log_{10} \frac{S_E}{p_{ref}^2 r_{ref}^2 t_{ref}} \]

Where \( p_{ref} \) is the reference pressure in water of 1 μPa, \( t_{ref} \) is reference time and \( r_{ref} \) is reference range. For the continuous sound sources such as ships, the estimated sound pressure level received by the animals at a particular point in space and time was estimated as
and for the transient sources such as underwater explosions and airguns, the sound exposure level is calculated from the energy source level

\[ SEL = SL_E - PL \text{ dB re} 1\mu\text{Pa}^2 \text{ s} . \]

A relation between SEL and SPL which is based on the time dispersion may be used to be able to compare them in the same metric. Estimation of time dispersion gives an insight about the size of averaging time window. The SEL-SPL difference is derived for a point source in the range independent environment [10]. This work has been extended for range dependent media and dipole sources (e.g. airguns).

Sound maps are generated for shipping, underwater explosions and airguns in the Dutch part of North Sea. For ships, the shipping density data from 2007 [11] is used. The ships are modelled as continuous point sources. The average source level for ships are calculated by the Wales and Heitmeyer formula\cite{4}. In Fig. 1, SL for a single ship is shown.

Underwater explosions generate transient sounds which can be described more accurately by the energy source level. The energy source level for underwater explosions is modelled by empirical approximations \cite{5}. The shock waves and the first and second bubble pulses are considered in the time domain waveform. Approximately 230 underwater explosions are taken into account from the period of 2010 and 2011.

Airguns are commonly used in an array during seismic surveys. The calculation of airgun signatures requires more sophisticated methods which take into account the bubble motion, gas pressure, mass transfer and optimization models \cite{6,7}. The airgun arrays are fired very close to sea surface. Thus, the reflected time signal from sea surface (ghost) contributes to time domain waveform. A new airgun signature algorithm (AGORA), which calculates the notional airgun signature (the time domain signature without the surface ghost) of each airgun in the array has been developed. The airgun signature with the ghost (a flat sea surface) is calculated by using a mirror image of source \cite{8}. In Fig. 2, the airgun signatures with and without the surface ghost are shown.
In this work, the notional airgun signatures are directly used in the propagation algorithm including the effect of vertical directivity on propagation. Sound maps of seismic surveys are generated for the seismic surveys in the North Sea between 2007 and 2008. A different airgun array layout is used for each seismic survey. The effect of vertical and horizontal directivity on propagation is taken into account.

As natural sound sources, wind and rain are considered. For the sheet sources such as rain and wind, analytical approach is used [9]. The wind speed and precipitation data for each month are obtained from the KNMI [Royal Netherlands Meteorological Institute]. Monthly averaged sound maps for rain may not provide a good insight. Thus, weekly, daily or hourly precipitation inputs are preferred for more accurate estimation.

## 3. PROPAGATION LOSS CALCULATION FOR SOUND MAPPING

Accuracy of propagation loss (PL) is significant in sound mapping as well as the other applications of underwater acoustics such as sonar performance models, acoustic communication, etc. Underwater acoustic propagation need to take into account many environmental factors such as variable water depth, sediment type, sound speed (profiles) etc. PL can be calculated by different methods and approximations [12]. The choice between these methods can be done depending on the frequency range, water depth and the variation of environmental parameters versus depth and range. For a sound map calculation involving many different sources, variable water depth and sediment type, the stability and computational speed of the propagation method needs also attention. The performance of different PL models were compared by Sertlek and Ainslie [9]. While, mode theory can provide accurate and stable results, Weston’s flux theory [13] provides faster analytic solutions without requiring advanced numerical algorithms. As a disadvantage, the depth dependence of PL is not considered precisely in the Weston’s flux integral. Mode theory requires long computational times especially for high frequencies, deep water and long range (for the range dependent waveguide) problems. We use a new approach that combines the advantages of both methods in this paper. A hybrid algorithm, SOPRANO(SOurce and PRopagation Algorithms for NOise Assessment), based on Weston’s flux and an incoherent mode sum [3] has been developed for shallow water problems which can provide fast results with a similar accuracy to mode theory solutions. The different source models with their far field directivity patterns can be also handled by
this approach. As a result, SOPRANO can be an alternative tool especially for the large scale problems, as sound mapping.

4. SOUND MAPS FOR THE NORTH SEA

In this work, sound maps are generated for the anthropogenic (shipping, underwater explosions, seismic surveys) and natural sources (wind, rain) at the center frequencies of 1/3 octave band for different receiver depths (1 m below sea surface, 1 m above seabed and middle water depth). These water layers reflect ecological extremes (e.g. surfacing dolphins and benthic fish) with also largely divergent acoustic conditions, while diving mammals and many pelagic fish species may often experience some sort of intermediate conditions. The spatial resolution of these maps can be specified based on the scale of movement of the study species and required precision of the estimated SEL or SPL. Also the nature of the source (e.g., whether a source is stochastic and incidental (e.g. explosions), or whether it is of a more continuous nature (e.g. shipping)), requires different acoustic metrics and temporal resolution.

For shipping noise, the average source level is modified based on the total number of ships per grid cell in the sound map. The source depth for all ships is assumed to be 4.5 m. The shipping density data from 2007 is only for the ships in the Dutch part of the North Sea. Other ships which are not in the economic zone of Netherland are not taken into account. These ships can be added when data become available. For the underwater explosions, source depth is assumed as 0.5 m above seabed. The same frequencies and receiver depths are used as ships. For the explosion maps, the time period for the averaging is critical due to the animal distribution. Weekly, monthly and annually averaged sound maps are generated for this purpose. However, if we would aim to depict the potential impact on direct behavioural responses, we would need to map each explosion separately. In Fig.3, an annual sound map for shipping (left) and explosions (right) are shown.

Figure 3. Annually averaged SPL map of shipping for 2007 (left). Sound map of explosions(right) (averaged over two(2010-2011) years). Frequency is 125 Hz. Receiver depth is 1 m.
These maps (in Fig.3) are generated by another propagation loss algorithm (*Aquarius*) which is also based on Weston’s flux formulations[11]. However, it uses different assumptions for depth and range dependency. Mode region effects are not included in these maps. The frequency is 125 Hz and the receiver depth is 1 m. The sediment type is assumed as a medium sand. Sound speed is a constant as 1500 m/s.

For seismic survey maps, the directivity of the source is taken into account. For airgun arrays, most of the source energy is at low frequencies. Thus, the frequency range in sound maps of seismic survey maps is different from that of other sound maps. Because of the computational limitations, these maps are calculated up to 1-3 kHz depending on the airgun array layout. The maps are generated for each airgun shot for the individual seismic survey maps. In Figure 4, a SEL map for a single shot is shown at 1 m receiver depth. The notional airgun signatures are calculated by AGORA for an airgun array with 17 airguns. The total volume of airgun array is 3333 in³. The firing pressure of each airgun is 2000 psi. The source depth is 5.5 m. SOPRANO is used to generate SEL map including vertical and horizontal directivity.

As can be seen from Figure 4, high SEL (maximum SEL is 173.5 dB re1μPa² s around 9-10 km) can be seen for each shot. Seismic surveys typically have short shot intervals such as once every 50-100 meters, thus a large number of shots (in thousands) can be observed in the seismic survey area. This means seismic surveys make one of the largest contributions to soundscape depending on the distribution of seismic surveys for each year [14]. When data on all seismic activities is available, the annually averaged sound maps for seismic surveys can be estimated.

5. CONCLUSIONS

The mathematical estimation of sound maps can provide insight into the distribution of sound intensity versus range, depth and frequency. The accurate and stable calculation of propagation loss and source model characteristics are critical in sound mapping. Each
source can contribute to a different extent to different frequency bands. For example, airguns have higher energy source levels at low frequencies (less than 1 kHz). On the other hand, the dependency of PL varies with frequency. These details can be critical for some animal species depending on their sensitivity or hearing and swimming depths. To understand and interpret these effects in more detail, the sound maps can be adapted for any animal species, providing existing knowledge of their sensitivity and behaviour. This paper describes maps created primarily for marine mammals and fish in the North Sea but could be developed for other areas and species, depending on the availability of local information. Such maps can be a useful indicator to inform regulators involved with marine spatial planning of seismic surveys, shipping routes and underwater explosions.

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Session 22

Synthetic Aperture Sonar: State-of-the-art

Organizers: Roy-Edgar Hansen and Daniel Brown
CHANGE DETECTION IN TOPOGRAPHIC STRUCTURES USING INTERFEROMETRIC SYNTHETIC APERTURE SONAR

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Abstract: Interferometric synthetic aperture sonar (SAS) is a sensor technology well suited for imaging and mapping of the seabed, providing high resolution and large area coverage at the same time. When repeated passes are made over the same area, the data can be used to detect changes at the seabed between the passes. Automated change detection may be useful in many areas within marine research, offshore oil and gas, and military applications.

A critical stage in change detection processing is coregistration of the images, that is, accurately positioning the image grid from the second pass on top of the image grid from the first pass such that image based techniques can be used to detect changes. Traditional navigational accuracy on towbodys and autonomous underwater vehicles (AUV) is not high enough to supply coregistered images without using data driven techniques (the sonar images themselves). Topographic structures may be a challenge in SAS imaging, potentially producing layover, shadowing, multiple reflections, defocusing and projection errors. This causes also difficulties in the coregistration process and the change detection process for repeat pass data.

In this paper, we consider a ridge outside Trondheim, Norway, containing coral reefs. Data were collected by a HUGIN AUV carrying an interferometric SAS in repeated passes December 2012 and December 2013. The ridge is a topographic structure with relatively large vertical variations where coral reefs are stationed. We perform a data exploratory analysis of the challenges involved in the SAS processing and in the coregistration, and show example images and bathymetries of the passes.

Keywords: Synthetic aperture sonar, Interferometry, Coregistration, Change detection
1. INTRODUCTION

Fig. 1: Left: The HUGIN HUS AUV equipped with the HISAS 1030 interferometric SAS. Right: The vehicle track over the ridge from the 2013-mission.

Synthetic aperture sonar (SAS) is a signal processing technique which increases the along-track resolution in sonar images by coherent combination of collected pings [1]. The field of SAS is small and very similar to its counterpart synthetic aperture radar (SAR) [2]. In recent years, SAS technology has, however, gained popularity in many applications related to high resolution imaging and mapping of the seabed.

Interferometry is a processing technique for estimating time delay differences (or absolute phase differences) between images [3] at different locations and/or at different times. For a SAS system with two arrays with a vertical baseline (displacement between sensors), interferometry can be used to estimate the seabed depth or the bathymetry [4].

Change detection is the process of finding changes of interest in images with a temporal difference (or baseline). This has potentially many applications in sonar and radar [5, 6]. Image based change detection relies on accurate coregistration of the data from the repeated passes, where the requirement is related to the theoretical image resolution. For typical high frequency SAS systems, the theoretical resolution is on the order of centimeters, while the global positioning accuracy on the platforms carrying the SAS systems are on the order of meters [7]. This implies that data-driven coregistration is a required step in change detection.

In December 2012 the Applied Underwater Robotics Laboratory (AUR-Lab) at the Norwegian University of Science and Technology (NTNU) and Norwegian Defence Research Establishment (FFI) organized a series of tests using ROV and AUV from RV Gunnerus in the Trondheim Fjord [8]. This included using FFI’s HUGIN HUS AUV equipped with interferometric SAS (see the left panel of Fig. 1) [9]. One year later, in December 2013, very similar tests were conducted using the same equipment. During these experiments, the AUV ran the same mission lines over a ridge containing corals (right panel of Fig. 1).

In this paper, we investigate sample SAS data from the two cruises in an exploratory manner. An interesting question is what the challenges are for using interferometric SAS on AUVs in detecting and monitoring changes in rough topography. In previous work, we have evaluated for noncoherent change detection with short and long temporal baselines [10], and coherent change detection with short temporal baselines [11]. Both studies were in relatively benign areas. In this paper we describe potential sources for errors and list potential techniques for coregistration of SAS data in rough terrain with large temporal baselines (one year).
2. DATA DESCRIPTION

Fig. 2: SAS image drawn on top of the bathymetry from the 2013-mission (Master).

We consider SAS data from the 2013-mission (referred to as Master) and the 2012-mission (referred to as Slave) of a 50 x 100 m scene on top of the ridge (see Fig. 1). Fig. 2 shows the Master SAS image relative to the estimated seabed depth and the vehicle tracks. Note the large depth variation for the vehicle tracks, more than 200 m depth at the deep end and around 50 m depth on top of the ridge. Since the vehicle follows the seabed on a smooth track with fairly constant altitude, severe heave and pitch will be induced.

The ocean environment is challenging both for vehicle operation and SAS data processing. The sound speed varies with depth (left panel of Fig. 3). It was also variations between the different mission lines within one mission, and between missions. We see that there is a sound speed gradient between the vehicle depth and the seabed depth (orange lines) in 2012, and less so in 2013. The right panels of Fig. 3 shows the vehicle position and attitude during the data acquisition for the SAS images. The vehicle speed was around 2 m/s. Note the particularly challenging depth and pitch variation over the length of the synthetic aperture. Note also that there is an average horizontal track direction difference between the passes.

Fig. 3: Left: Sound speed profile for the 2012 and 2013 mission. Right: Estimated vehicle navigation (position and attitude) for the data used in the synthetic aperture formation.
3. SAS IMAGE AND INTERFEROMETRY PROCESSING

The generalized processing flow for interferometric SAS processing is illustrated in Fig. 4. SAS imaging is essentially to transform sonar data from time to space, where accurate knowledge of the ocean environment (the sound speed), the seabed topography (the geometry), and the vehicle track (the element positions in the synthetic aperture) must be obtained [9]. A critical stage in successful SAS is to use the sonar data itself for navigation correction (also known as micronavigation). This is needed to meet the required accuracy in positioning the elements along the synthetic aperture. For nonstraight vehicle tracks (or nonlinear synthetic apertures), accurate knowledge about the seabed topography is needed. This is a well known result [9] which is particularly important for operating SAS systems on small vehicles close to the seabed in rough terrain.

Fig. 5 shows a fusion of the Master SAS image (as brightness) and the seabed depth estimate from interferometry (color coded). The SAS image is constructed using the backprojection algorithm in three dimensions and the theoretical horizontal resolution is around 3 x 3 cm. We have done the interferometry processing using an absolute phase difference estimator based on the cross correlation for wideband systems [4], with a coherence estimation window of size 18 x 18 cm. The scene contains a topographic structure that is elevated around 12 m. The textured areas are corals and the smooth areas are a sandy type seabed. The vehicle altitude is around 20 m higher than the highest point. The black area around along-track position 80 m, cross track position 65 m is shadow cast by the structure.

Fig. 4: SAS image and interferometry processing flow.

Fig. 5: Fusion of Master SAS image (brightness) and SAS bathymetry (color).
4. REPEAT PASS ANALYSIS

A critical process in repeat pass analysis is coregistration of the data. The horizontal position accuracy for state-of-the-art Doppler velocity logger aided inertial navigation for AUVs is on the order of meters running in typical lawn mower pattern missions [7]. This is much poorer than the grid resolution of the data (see the previous section). Hence, data driven techniques (that is, using the sonar data itself) are required for accurate coregistration. Note that this is a fundamental difference between SAS and SAR. In SAR, one often assumes and relies on the navigational solution being very accurate, and coregistration for correcting the tracks in large scale is generally not needed.

Fig. 6 shows an overview of repeat pass processing for change detection. The data from the two passes (master and slave) are conditioned and gridded in the same earth fixed coordinate frame. Then data driven coregistration is applied. There are essentially three different levels of grid based change detection:

1. **Coherent image based change detection**
   uses the single look complex images formed at each pass (see Fig. 4), and estimates differences in phase and amplitude similar as interferometry does in seabed depth estimation [5, 3]. This technique allows for detecting changes not visible in the signal amplitude. For this technique to work, speckle in the images must be preserved over the repeated passes. This implies that the vehicle tracks must be very similar, and the seabed must not change [12]. In addition, the images must be coregistered within a fraction (one tenth) of the theoretical resolution.

2. **Noncoherent image based change detection**
   uses the signal amplitude in the image. This technique is better suited when the speckle in the images has changed such that coherent techniques cannot be used, or amplitude changes are sufficient. If the speckle content has changed, despeckling of the images should be applied before data driven coregistration. Feature extraction and mapping using the SURF technique is one candidate to estimate the warping function [10]. This technique requires that the sonar look direction on the imaging scene must be fairly similar such that the texture in the images are similar (typically half of the beamwidth of the sonar elements or less). The required accuracy for coregistration is on the order of the theoretical resolution.

3. **Bathymetry based change detection**
   uses the estimated seabed depth from single pass interferometry processing. This technique can be used if the sonar look direction on the scene is large (e.g. opposite views). The required accuracy for coregistration becomes comparable to the resolution of the horizontal resolution of the bathymetry maps.

Obviously, if the requirements for coherent change detection is met, noncoherent and bathymetry based change detection can also be performed.
There are different strategies in the preprocessing of the data in change detection. For noncoherent or bathymetry based change detection, a potential candidate is to run the full processing including geocoding (all the parts in the flowchart in Fig. 4). Then run data driven coregistration. In this work, we have chosen to use the backprojection algorithm with common earth fixed coordinates for the master and slave mission lines (common imaging plane in Fig. 4), similar as described in [11]. This technique allows better to assign the sources for deformation and misregistration. After imaging, we ran the feature based coregistration algorithm which estimated a coarse shift of around 2.6 x 1.2 m between the two images. We then ran re-imaging of the slave image (see Fig. 6). The master and slave SAS images are shown in Fig. 7. We see that the texture features in the images are very similar. In Fig. 8 we have zoomed in to a 8 x 4 m area of the master and slave image. We see variations in the image quality and in the shadow mapping and that the fine scale variability related to speckle is not preserved. This implies that coherent change detection will be very difficult or impossible.

Fig. 7: SAS images of the two passes. Upper: Master from 2013. Lower: Slave from 2012.

Fig. 8: 8 x 4 m zoomed area of SAS master (left) and slave (right) image.
In order to investigate how well the data are suited for use in change detection, we performed a magnitude based local shift estimation. We applied simple despeckling and downsampling, and estimated the normalized cross correlation coefficient using a 42 x 42 cm correlation window and a 302 x 302 cm search window (tolerating up to $\pm 150$ cm shift in both dimensions). For each local correlation function, the peak value, x-shift, and y-shift of the peak was stored. The upper panel of Fig. 9 shows the local peak correlation coefficients. For small number of samples, the correlation coefficient suffers from bias at low values [3] which results in a lower threshold at around value 0.3, where the value should not be trusted. When we applied the same method on the complex images (including phase), we obtained no high values. Given that the search window is chosen intelligently, and that dilation and rotation effects does not cause decorrelation, this can be used to map the preservation of texture between the passes. We see that the correlation coefficient is high in the textured areas in the images (see Fig. 7). This indicates that the texture is preserved in the areas around the elevated structure and where there are corals. In the smooth areas, the correlation coefficient is low.

The lower panels of Fig. 9 show the estimated x-shift (left) and y-shift (right) between the two images. We have thresholded the data at correlation coefficient 0.5. Dark blue pixels indicate invalid values. One pixel equals 2 cm. There are consistent local shift estimates that indicate a nonlinear warping function. There is a large misregistration along-track at around $x = 20 - 30$ m. This may be related to inaccurate estimates or incorrect assumptions of the topography or the track (note the pitch change in Fig. 3). We also see a misregistration cross-track that can be related to topography errors, or induced by incorrect use of the sound speed profiles. This indicates that a simple warping function including global shift, scaling and rotation will not be sufficient.

The choice of imaging plane is critical and should be sufficiently smooth to avoid render errors, and close enough to the true seabed map in the imaging scene to optimize performance. It should be common, based on both the master and slave data, and optimized for best fit for both tracks. Running iterative processing may lead to improved performance.

![Fig. 9: Upper: Peak values of local magnitude cross correlation between images. Lower: Estimated shift along-track, the x-axis (left) and cross-track, the y-axis (right).](image-url)
5. SUMMARY

Interferometric synthetic aperture sonar carried by an underwater vehicle is a well suited tool for detailed imaging and mapping of the seabed, providing high resolution and large area coverage rate simultaneously. When repeated passes of data are collected, the image products can be used in detecting changes over time. Grid based change detection requires that the SAS data quality is high and that the images are accurately coregistered. In areas of rough topography and variable ocean environments, all the processing stages becomes more challenging. Successful SAS processing becomes more difficult and interferometry is more demanding. The warping function needed may be nonlinear to encapsulate the misregistration. Automated change detection in rough terrain is work in progress, and we will continue our work on this.

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REFERENCES

COMPARISON OF FUSION APPROACHES FOR THE DISPLACED PHASE CENTRE ANTENNA METHOD

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Abstract: Although autonomous underwater vehicles (AUVs) with synthetic aperture sonar (SAS) have been employed by navies successfully for several years now, further improving its robustness is nonetheless an important research topic. For synthetic aperture imagery the theoretical azimuth-resolution is enhanced compared to real aperture imagery, but only if the sonar signals are processed coherently. The prerequisites for that are ping-to-ping coherence and adequate motion estimation. Typical problems that jeopardise these requirements are multipath effects, noise, or uncompensated sonar motion. This paper focuses on the motion estimation, which is performed by fusing data from the inertial motion unit and the sonar itself. The latter estimates ping-to-ping displacements based on the Displaced Phase Centre Antenna (DPCA) method that uses sonar data from overlapping phase centres. In cases where DPCA provides inadequate accuracy to enable full resolution SAS imaging, incorporating additional techniques can be beneficial. The time delay estimation inside the DPCA method is typically performed many times per ping. The vast amount of time delay measurements generated provides access to the motion estimation accuracy, e.g. sudden jumps from ping-to-ping may be easily filtered out. Knowing the estimation accuracy at all times also enables to fuse information when several SAS arrays are in use. In this paper data are analysed from the SeaOtter AUV fitted with the Vision1200 dual-sided interferometric SAS, a long range autonomous mine hunting system. Four SAS arrays are included in the analysis and different fusion techniques are compared. In the analysis the relevance of estimated motion between port and starboard arrays is regarded in particular.

Keywords: Synthetic aperture sonar, Displaced Phase Centre Antenna, Correlation
1. INTRODUCTION

Synthetic aperture sonar (SAS) processing coherently combines data from multiple sonar pings, with the objective to improve resolution, area coverage rate and signal-to-noise ratio (SNR). Ping-to-ping coherence and accurate sensor positioning are prerequisites for a successful SAS. The latter is the most challenging [1] and means that for each signal the corresponding transmitter and receiver position should be known with an accuracy of a fraction of the wavelength. Since navigation sensors alone generally do not suffice, the sonar itself is included for motion estimation.

Motion estimation with the sonar is referred to as micro-navigation, and is based on sonar data obtained from overlapping elements. When the sonar moves less than half an array length per sonar ping, some elements, called overlapping phase centres, can be assumed to comprise the same signals, because the geometry dictates that their signal travel times are sufficiently similar. In the case of deviation from the assumed geometry (usually a straight line) the differences in these signals can be exploited for motion estimation, which is referred to as the displaced phase centre antenna (DPCA) method. DPCA can estimate surge from which overlapping phase centres correlate best. When the overlap is large enough, it is beneficial for the estimation to beamform the phase centres increase SNR and directivity. Furthermore it can estimate sway, heave, yaw and pitch, but only sway and heave sufficiently accurately. In the SAS processing implementation signal correlation is performed for several different slant-range windows and for each overlapping phase centre, which results in many available time delay estimates per ping.

Usually the DPCA functions adequately, but in difficult situations erroneous motion estimation leads to defocusing and high sidelobes. This paper proposes robust motion estimation by using information from the arrays mounted on both sides of the autonomous underwater vehicle (AUV), as shown in Fig. 1. Section 2 describes two fusion approaches. In Section 3 they are compared and in Section 4 a statistical analysis shows how beneficial the fusion is. A conclusion is given in Section 5.

Fig.1: Survey geometry for SeaOtter fitted with the dual-frequency dual-sided interferometric synthetic aperture sonar Vision1200.
2. DPCA FUSION APPROACHES FOR MULTIPLE ARRAYS

The SAS processing used for the analysis here is based on time-domain imaging preceded by motion estimation based on navigation data and the DPCA method. Both SAS imaging and motion estimation is performed in the 3D Cartesian coordinates along-track $x$, ground-range, $y$ and height $z$. The ping-to-ping correlation in DPCA first provides surge and slant-range sway estimates for each range window and for each beamformed phase centre. The slant-range sway estimates are then converted to sway and heave estimates. The 3D approach facilitates adapted processing for non-flat sea floors and interferometric SAS processing. It also helps to fuse information of the different SAS arrays mounted on the AUV.

Motion estimation is a crucial step in the formation of SAS imagery. The translational motion estimated with the DPCA method should be and often is as accurate as a fraction of the wavelength. This means millimetre accuracy for the Vision1200 SAS. Problems that can affect the motion estimation are manifold. The accuracy of the estimation depends on the ping-to-ping coherence [1] and can be jeopardised additionally by ambiguities. Furthermore, the following assumptions in (the implemented) DPCA are not always valid:

* The sound speed is constant. If this is not the case, conversion from time to range is erroneous. When the sound speed varies with depth, refraction changes the direction of arrival of the signal, and hence the conversion from slant-range-sway to sway and heave.

* The sea bottom profile is known. When the assumed profile is assumed flat, errors in the signal’s direction of arrival may lead to errors in heave and sway.

* Many stationary point scatterers on the sea floor in broadside direction form the signal received. The assumption is violated when a dominant target is present, which causes the estimated sway direction to deviate from the intended direction, i.e. it biases towards the strong target. Stationarity is violated in the presence of moving targets, e.g. fish.

Motion estimation errors that occur on one side of the AUV do not in general arise on the other side. Based on this principle a fusion algorithm for the port and starboard arrays is investigated here. Two fusion approaches are investigated, one implemented at time delay estimation level and one at motion level. The latter uses a Kalman filter [2] applied to the translational state vector surge, sway and heave of the four arrays as in [3], [4]. The former uses a non-linear least squares technique [5] that estimates the sway and heave estimates that fit best to all the time delay estimates of the four arrays. For both fusion techniques sway and heave motion is calculated at a common reference point, in this case where the inertial motion unit is. The two approaches are explained using an example that was recorded with the SeaOtter AUV off the German island Rügen in the Baltic sea on October 25, 2013. The AUV was sailing in mid-water and the water depth was around 20 m. 110 SAS pings are processed for the high frequency band centred at 150 kHz, and for a ground-range between 20 and 100 m. This particular example showed better image quality on the starboard arrays than on the port arrays. The DPCA estimates are shown for one ping with a circle for each range bin in Fig. 2a. The circles show a different behaviour versus range for each array due to the fact that their slant-range plane differs, for example due to orientation. Heave and sway are estimated by fitting (the thin) curves through the circles, for each array separately, resulting in the curves in Fig. 2c-d. The heave and sway that fits best to the DPCA estimates of all four arrays are plotted in Fig. 2c-d (black circles). The non-linear least squares fit error for all four arrays versus heave and sway is shown in Fig. 2b. The curves corresponding to the optimum are the thick ones in Fig. 2c.
Fig. 2: Non-linear least squares fit approach to fuse four arrays.

Fig. 2a shows that some DPCA estimates deviate, and are likely wrong. It is when too many estimates are wrong that the motion estimation fails. Fig. 2b shows that sway is more easily estimated than heave, but the sway accuracy requirements are accordingly. Fig. 2c shows that the sway estimates of all arrays are within a millimetre and Fig. 2d shows that the heave estimates are within a centimetre. There is also one outlier at ping 39. The non-linear least squares fit approach provides reasonable estimates, where it appears to trust the starboard array data more. Two outliers in heave appear here too.

The Kalman filter approach has the advantage that the output, estimated by a prediction step based on the previous ping and a correction step based on the four array estimates, can still be different for each array. For example, when the measurement error is the same for each antenna, it is better to trust the own array estimate more than the array on the opposite side. The Kalman measurement noise matrix is set here by a combination of measurement error and transferability (i.e. how alike are port and starboard side). The DPCA estimation error is often calculated by the theoretical minimum: the Cramér–Rao lower bound (CRLB), which follows from the ping-to-ping correlation coefficients and is plotted in Fig. 3a. This value was found to significantly underestimate the motion estimation error, because it ignores a number of effects, e.g. bottom profiles. Two alternative motion error estimates were analysed in this study. The one in Fig. 3b is calculated with the difference between DPCA estimates at ping \( p \) compared to the mean of the DPCA estimates at ping \( p-1 \) and ping \( p+1 \). Fig. 3c shows the other alternative, the non-linear least squares fit error. The Kalman filter results for the port bottom array using a measurement error matrix based on the ping-to-ping differences is plotted in Fig. 3d-e.
Fig. 3: Kalman filter approach to fuse four arrays.

Fig. 3a shows that the CRLB for this example is always very small. If the true motion error were that small, no improvement would be possible. Fig. 3b and c nevertheless show a different result, and suggest that improvements are possible. Fig. 3b shows a clear error peak at ping 39, where the sway and heave jump occurred. Fig. 3c shows the fit errors for each array. A much clearer (overall) difference between port and starboard is visible with this error estimate. The Kalman filter results shown in Fig. d-e demonstrate that the presumable jump error at ping 39 can be easily corrected.

3. COMPARISON OF FUSION APPROACHES

The SAS image that was generated with the original motion estimation of one array is compared to the SAS images based on the two fusion approaches described in the previous section. The images are generated with SAS data of the port bottom array. In addition, the same image is generated based on the motion estimates of the starboard array. Figure 4 shows the four SAS images, which have a dynamic range of 40 dB and a horizontal range axis from 20 to 100 m and a 30 m wide vertical cross-range axis. The top image shows some defocus problems in the bright area, which turns out much better for both fusion approaches. The bottom image shows that the starboard motion estimates (magenta curves in 2d-e) provide similar results to the fusion approaches. This is explained by the fact that the motion estimation is better on the starboard side in this example. The reason for that is the somewhat lower correlation due to the patches with low bottom scatter (dark parts in the image). This can be seen with the CRLB (Fig. 3a) and the fit error (Fig. 3c), and is therefore noticed by the fusion approaches. The three image quality metrics that have been calculated for the images, maximum, contrast and signal excess (maximum-median) also indicate that fusion is beneficial, and that exchanging of the port and starboard side estimates is undeniably possible.
Fig. 4: SAS images of an area near Rügen, northern Germany.

Original
max = 109.4 dB  
contrast = 2.8 dB  
SE = 23.5 dB

NLLS Fit
max = 110.5 dB  
contrast = 2.9 dB  
SE = 23.9 dB

Kalman
max = 111.2 dB  
contrast = 2.9 dB  
SE = 23.7 dB

PS Swap
max = 111.2 dB  
contrast = 3.0 dB  
SE = 24.1 dB
4. STATISTICAL BENEFIT ANALYSIS FOR PORT AND STARBOARD FUSION

The Kalman filter based fusion of the four arrays has been tested on 34 datasets. The datasets were collected in different areas, and at different times, and they include data from the two frequency bands (centred at 75 and 150 kHz) that Vision1200 is operated at. In this section statistical results are shown for the differences between port and starboard estimates to provide insight as to how well fusion could work. When the differences in sway and heave are usually small, one could potentially fuse well. First the offset is considered. As shown in [6], an offset in either sway or heave can occur when the arrays are not mounted perfectly parallel on the AUV. The offset for sway for the example dataset is plotted in Fig. 5a and the offset values for all datasets are plotted in Fig. 5b.

![Fig. 5: (a) Sway differences between port and starboard and between top and bottom for the example. (b) Heave and sway offset for all datasets.](image)

The aim is to establish with many measurements whether the arrays are mounted to the AUV sufficiently accurately, or that compensation is required. If the offset in sway $\Delta s$ is present consistently, it is likely explained by a yaw difference. For the top and bottom arrays no offset was found, but Fig. 5b shows a clear small offset in sway and a somewhat larger offset in heave. The median offset value is 0.2 mm for sway -1.3 mm for heave, which means it is negligible and motion estimates should be exchangeable between port and starboard side.

The second statistical result from the 34 datasets is the similarity of port and starboard estimates. If they are sufficiently similar, exchange or fusion is possible. The similarity criterion is that the sway estimates have to be within a fraction of the wavelength and the heave estimates within a wavelength, which was supported by simulations and theory. Fig. 6a shows the similarity in terms of standard deviation of the port-starboard difference, for both sway and heave. The standard deviation is indeed small for heave and even smaller for sway, due to the geometrical fact that sway accuracy is higher than heave accuracy. Fig. 6b shows the distribution function of the two, which suggests that over 90% of the estimates are sufficiently similar to enable port and starboard fusion. In the near future the data will be reprocessed to establish statistically how much gain can be expected from the fusion approaches.
5. CONCLUSION

In this paper two approaches to fuse motion estimation from several SAS arrays were described and applied to an experimental dataset. It is shown that such fusion robustifies the SAS processing by mitigating motion errors. In addition, a statistical analysis on a multitude of SAS datasets was performed and presented. The analysis shows that port and starboard motion estimation data are almost always compatible, and can therefore be exchanged or fused. The sensors, mounting and system component positioning are proven adequate for fusion of motion estimation of the four Vision1200 arrays on the SeaOtter AUV.

REFERENCES


ALTERNATIVE SAS PROCESSING FOR GAS SEEP DETECTION

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Abstract: An unknown amount of the greenhouse gas methane is continuously seeping into the oceans, impacting marine life and potentially reaching the atmosphere. The introduction of subsea CO₂ storage further increases the need for accurate monitoring methods to ensure that potential leaks are detected. Gas seeps in the water column appear as characteristic “flares” in single- and multibeam sonar images due to the high contrast in acoustic impedance between water and gas. We investigate the potential of using synthetic aperture sonar (SAS) for seep detection. SAS is known to provide high quality seafloor images with significantly improved azimuth resolution. However, standard (coherent) SAS is not suited for gas seep imaging since the required assumption of temporal stationarity is violated. We propose an alternative processing scheme where we produce two SAS images - one standard SAS image and one where the images from individual pings have been combined noncoherently to preserve the intensity of a gas seep. The difference in mean pixel intensity reveals the presence of a seep. We collected data from two seep locations in the North Sea where shallow gas is escaping through cracked cement well casings. The seep was imaged using the HISAS 1030 sonar carried by the Hugin AUV. We show how the proposed processing scheme can be used to detect and accurately localize even a modest seep.

Keywords: Seep detection, Sonar, Seafloor mapping, SAS, noncoherent SAS
1. INTRODUCTION

In recent decades, there has been an increasing interest in monitoring the seafloor in order to detect gas seepage into the oceans. Gas seeps may arise as a result of human activity such as oil and gas production or subsea CO₂ storage. They may also be natural seeps caused by migration of gas from shallow or deep sources through the seabed. Natural seepage of methane and CO₂ is common in many regions where geological structures such as faults or salt domes allow these greenhouse gases to migrate upwards and eventually enter the water column and potentially also the atmosphere [1].

Gas bubbles in water are strong acoustic scatterers, making sonar an excellent tool for detecting their presence. Numerous studies demonstrate the successful use of multibeam and singlebeam sonars for seep imaging [2] [3] [4]. Several recent works also demonstrate the use of split-beam echosounders to derive specific properties of oil and gas seeps in addition to merely detecting their presence [5, 6]. Leighton et. al have proposed the use of passive sonar to detect and locate the presence of gas bubbles by their acoustic emissions [7].

Synthetic aperture sonar (SAS) has the advantage of offering seafloor images with significantly higher imaging resolution than conventional sonar methods such as single- and multibeam sonar [8]. A SAS is typically carried by a towfish or an AUV and operates near the seafloor for optimal image quality. This is a strong advantage when searching for seeps in deep waters. However, a plume of rising bubbles is poorly imaged by a standard SAS system because it violates the required assumption of temporal stationarity. The non-stationarity of rising bubbles causes a local loss in image intensity, making a seep difficult or impossible to detect.

We propose an alternative SAS processing scheme aimed at detecting and accurately localizing gas seeps at the seafloor. In standard, coherent SAS, a large aperture is synthesized through coherent combination of focused images from multiple along-track pings. An alternative approach is to form a SAS image through noncoherent combination of individual images. This partially noncoherent approach is sub-optimal in terms of image resolution, but ensures that the image intensity is preserved in the presence of a seep. These two resulting SAS images have similar characteristics after dedicated processing to account for differences in amplitude and resolution, except in the presence of a seep. Because an active gas seep has different characteristics in the two images, they can be combined in order to maximize seep detection ability.

2. ALTERNATIVE SAS PROCESSING SCHEME

In SAS imaging, a long aperture is synthesized by allowing the sonar to travel along a line, illuminating the same region on the seafloor with several pings. A high-resolution SAS image is formed by combining the data from multiple pings [8]. Usually, the sonar has a multi-element receiver with $M$ elements, and all elements in all pings form the synthetic aperture. In standard SAS processing, the sea bottom reflectivity $SAS(i, j)$ of pixel $(i, j)$ is estimated by coherently combining the complex transmitter-receiver data $s_n(i, j)$ of all elements $n$ in the synthetic aperture. Since the data from one ping is independent of all other pings, a practical implementation is to form a focused image from the $M$ elements in each ping, and combine these images to form the SAS image:

\[
SAS(i, j) = \frac{1}{N} \sum_{n=1}^{N} s_n(i, j) = \frac{1}{P} \sum_{p=1}^{P} \frac{1}{M} \sum_{m=1}^{M} s_{p,m}(i, j),
\]
Figure 1: Left: The HUGIN AUV during recovery after successfully completing its data acquisition mission. The HISAS consists of an interferometric SAS mounted on each side of HUGIN. Right: Illustration of SAS image formation.

where \( P \) is the number of pings, \( N = P \times M \), and \( s_{p,m}(i,j) \) is the transmitter-receiver data for element \( m \) in ping \( p \), weighted and delayed to focus at pixel \((i,j)\). The concept is illustrated by Figure 1 (right).

SAS offers a significant increase in azimuth resolution compared to conventional sonar methods, as well as the desirable property of range-independent resolution. However, coherent integration over several along-track pings requires that the scene is stationary over the acquisition period. This is not the case for rising bubbles. The result is that the target strength of a bubble plume is under-estimated.

An alternative approach which is better suited for gas seep imaging is to combine the individual complex-valued images noncoherently to form what we will refer to here as a noncoherent SAS image, denoted \( SAS_{\text{mod}} \):

\[
SAS_{\text{mod}}(i,j) = \frac{1}{P} \sum_{p=1}^{P} \left| \frac{1}{M} \sum_{m=1}^{M} s_{p,m}(i,j) \right|^2.
\]  

Since we assume that the rising bubbles can be considered stationary for a single ping, the interim single-ping images (sum over \( M \) elements in (2)) are formed coherently. For the long acquisition period of the entire synthetic array, rising bubbles cannot be assumed stationary, and noncoherent summation is necessary to preserve the image intensity. This partially noncoherent approach is related but not identical to fully noncoherent SAS as described in e.g. [9] and [10].

The noncoherent SAS image, \( SAS_{\text{mod}} \), differs from the coherent SAS image in several ways. The azimuth resolution is degraded and becomes range-dependent in the noncoherent image. The appearance of speckle arising from constructive and destructive interference between multiple scatterers within a resolution cell, is reduced as a result of noncoherent processing. There is also a difference in image intensity for two reasons; increased speckle intensity in the noncoherent image as a result of envelope processing, and reduced processing gain from approximately \( N \) in the coherent image to \( N^\alpha \), where \( 0 < \alpha < 1 \), in the noncoherent image. Most importantly in the context of seep detection, the \( SAS_{\text{mod}} \) image preserves the backscattered energy from a time-varying process such as rising bubbles.

Figure 2 illustrates the expected response from a persistent scatterer such as a rock at the seafloor, and a gas seep, after amplitude normalization to account for differences in the mean speckle level. The response from a point scatterer is narrower and slightly stronger in the coherent image due to the superior azimuth resolution and coherent processing gain. We expect
Figure 2: Expected image intensity when imaging a point scatterer and a gas seep at the seafloor using coherent and noncoherent SAS. The point scatterer is expected to be narrower and slightly stronger in the coherent SAS image due to superior image resolution and coherent processing gain. The response from the gas region is expected to be low in the coherent SAS image compared to the noncoherent SAS image, because of the non-stationarity of rising bubbles.

After processing, the coherently and noncoherently synthesized SAS images display similar properties except in the presence of a time varying process such as a seep. The difference in local mean intensity between the two processed SAS images reveals the presence and location of a seep. Denoting the processed images $SAS^P$ and $SAS^P_{mod}$, we form the difference image as

$$SAS_{diff} = |SAS^P_{mod} - |SAS^P||.$$  

3. RESULTS AND DISCUSSION

We collected experimental data using the HUGIN AUV (Figure 1, left) during a sea trial in the North Sea in 2011. The seafloor in the area is flat and consists mainly of sand. Data was collected at two abandoned wells leaking small amounts of shallow gas. The HUGIN AUV is equipped with an interferometric SAS [11] with a center frequency of 100 kHz and a 30 kHz bandwidth.

Figure 3 (upper left) shows a coherent SAS image of a $50 \times 50$ m region of flat, sandy seafloor. The image appears grainy due to speckle. The location of a modest seep is indicated
Figure 3: Upper left: Coherent SAS image of a sandy seafloor with the location of a small gas seep indicated. Upper right: Noncoherent SAS image of the same region. Lower image: Difference image between coherent and noncoherent SAS. The seep is visible as a bright spot with a difference of about 25 dB relative to the background difference.

by a white ellipse. The image intensity in the seep area is slightly higher than in the speckle background, but this can easily be interpreted as rocks, shells or other highly reflecting objects. ROV inspection of the seep site revealed an abundance of shells gathered around the seep. The noncoherent SAS image in Figure 3 (upper right) resembles the coherent SAS image but appears smoothed in the along-track direction due to the reduced azimuth resolution. The speckle pattern is reduced through envelope processing of single ping images. The seep is visible as the area with highest image intensity. Although more visible in the noncoherent than the coherent SAS image, the seep is difficult to detect in either of these images alone.

The difference image, $SAS_{diff}$, is shown in Figure 3 (lower image), displayed in a dB scale and normalized by its mean value. Here, the seep location is clearly visible as a region with a large difference (about 25 dB) compared to the background. The difference in the sandy regions dominated by speckle is close to zero after amplitude normalization. A few strong scatterers (shells or rocks) also give rise to a significant difference between the two SAS images. While they represent potential sources of false alarms, the difference at these locations is about 5-7 dB below the difference at the seep location.
Figure 4: Cross-section through the gas region (upper left) and a rock (upper right), for the coherent and noncoherent SAS images prior to filtering. The difference in image intensity is about 9 dB at the seep location. The intensity at the rock is slightly higher in the coherent SAS image (by 0.9 dB). The lower plots show the same cross-sections after low-pass filtering.

Figure 4 shows a cross-section through the two SAS images in the along-track direction, at the seep location (upper left) and at a rock in a different area (upper right). At the seep location, the intensity of the noncoherent SAS image is about 9 dB above that of the coherent SAS image. The difference in peak response at the rock location, as shown by the cross-section in Figure 4 (upper right) is about 0.9 dB and comparatively small. The narrower and slightly stronger response in the coherent image is expected and in accordance with the illustration in Figure 2. Low pass filtering using a filter which is larger than the resolution cell of the noncoherent image is required in order to avoid false alarms in the presence of small, strong scatterers. Without filtering, the difference in peak image intensity at the rock is small (0.9 dB), but the difference in image intensity is large to both sides of the peak due to the narrower response of the coherent image.

The lower plots in Figure 4 show the same cross-sections after filtering using a 70 cm × 70 cm Gaussian low pass filter. The response in the gas region is about 8 dB, similar to the unfiltered case. The response from the rock is significantly reduced as a result of low pass filtering over a large region, and the difference after filtering is about 2 dB. The difference of about 1 dB over the entire along track distance is a result of non-perfect amplitude normalization of the images.
Figure 5: Upper left: Coherent SAS image of a gas seep located on a sandy seafloor. Upper right: Noncoherent SAS image of the same region. Lower image: Difference image between the coherent and noncoherent SAS images, with the seep detected as a bright spot with a difference of about 32 dB relative to the background.

An example from another leaking well is shown in Figure 5. In this case a trained eye may recognize the seep in the SAS images alone because of its flare-like shape. The difference image 5 (lower image) confirms the location of the seep.

In the examples presented above, the seeps are located on a flat, relatively uniform seafloor. In terms of CO$_2$ storage this is a typical case. False alarms may occur in situations where coherent SAS is challenging, including regions with extreme topography or noisy environments. The proposed algorithm requires sufficient coherence over the synthetic aperture to successfully form a high-quality coherent SAS image. By overlaying the suspected seep locations directly on the high-resolution coherent SAS image, false alarms can be interpreted and in many cases ruled out.

4. CONCLUSIONS

An active gas seep is poorly imaged by standard, coherent SAS because of the non-stationarity of rising bubbles. Coherent combination of multiple pings causes a local loss in image intensity
at the seep location. Noncoherent combination of single-ping images preserves the backscattered energy at a seep location. We propose an algorithm which combines these two SAS images to detect and localize a seep. Experimental results show that even a very modest seep can be detected. False alarms may occur in regions where standard SAS suffers because of a lack of coherence over the synthetic aperture.

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Session 23

Tank Experiments

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CALIBRATION OF ULTRASOUND TRANSDUCER HEADS USING SHORT PREPROCESSED ULTRASONIC PULSES

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Abstract: In a previous work the authors have presented a method for the generation of short ultrasonic pulses in medium size tanks. In this work an application of this method for the calibration of ultrasound transducer heads will be compared to typical calibration methods. The Laboratory of Underwater Acoustic Measurements at IACM-FORTH, Crete (Greece), owns a medium size water tank (3m x 1.5m x 1.3m) which is mainly used for reflection and calibration experiments. The transducers mainly used in these experiments are piezoelectric elements with central frequencies around 500 kHz and bandwidths of 200 kHz. The beamwidths of the narrow beams depend on the transducer head sizes. The manufacturer of the transducers does not provide calibration charts and documents for these instruments. Two different transducer heads, one with small diameter and one with large diameter, are calibrated using a conventional method and the one proposed here through the use of preprocessed ultrasonic pulses. Calibration was performed by the two methods at different distances using a calibrated reference hydrophone. Results from the two methods are compared and conclusions are drawn regarding the applicability of the new proposed method at calibration experiments.

Keywords: Tank experiment, Calibration, Pulse correction
1. INTRODUCTION

A projector calibration is a measurement of its transmitting voltage response, [1]. It is defined as the ratio of the apparent sound pressure at a distance of one meter, in a specified direction from the effective acoustic centre of the transducer, to the voltage applied across the electrical input terminals and it is measured in dB. Such measurements assume free-field conditions. A free-field is a homogeneous isotropic medium free from boundaries. Since a perfect free-field is never achieved, various practical means (pulsed sound, sound absorbers etc.) are used to counteract the absence of free-filed. In laboratory experiments performed in medium size tanks, short ultrasonic pulses and pulsetrains are often used for calibration experiments. The duration of the pulse depends mainly on the size of the tank and the frequency of the pulse. For low frequencies and small tank sizes the duration of such wave trains has to be small to avoid incoming reflections thus making it impossible to have ample cycles of the transmitted frequency. As a result the steady state of the received signal (necessary for measuring the amplitude) is not apparent or well defined due to the distortions caused by the presence of transients. In such cases it is desirable to have no transients on the recorded signal. In a previous work [2], the authors have applied a method for the generation of short ultrasonic pulses in medium size tanks, [3-6]. In this work, it is examined if this method can be used for projector calibration. Two piezoelectric projectors with different diameters were calibrated using both a standard secondary calibration method and also using the calibration method mentioned above. The results are compared and conclusions are drawn based on the applicability of the second method to calibration experiments.

2. PROJECTOR CALIBRATION (TYPICAL SECONDARY METHOD)

The Laboratory of Underwater Acoustic Measurements at IACM-FORTH, Crete (Greece) uses several pairs of ultrasonic immersion transducer heads of different sizes. These transducers are piezoelectric elements with central frequencies around 500 kHz and bandwidths of 200 kHz. The manufacturer of the transducers does not provide calibration charts and documents for these instruments. Two of the heads, one with small (V318, Panametrics) and one with large (V389, Panametrics) diameter were calibrated using a standard secondary calibration method.

In particular each transducer was placed inside the tank facing along its main axis a reference hydrophone with known sensitivity (TC4038, Reson) and usable frequency range from 10 kHz to 800 kHz. The transducer was connected to a wave generator through a linear power amplifier (Fig. 1). A sinusoidal wave train consisting of 20 was successively spanning a frequency range from 300 kHz to 700 kHz (with a 10 kHz step). The transmitted amplitude of the projector was kept constant at $V_p=150$ Volts (p-p), was received by the hydrophone and was recorded on a PC using a high speed Data Acquisition Card (DAC). In successive experiments, the distance between the projector and the receiver $d$, was changed with a step of 1 cm from 20 cm to 50 cm. The steady state amplitude $V_h$ of each signal arriving at the hydrophone was then calculated. In Fig. 2 a typical signal corresponding to a frequency of 500 kHz and separation distance $d=25$ cm, is shown. In this figure, the red part of the signal was considered as the steady state and was used for the estimation of $V_h$. Knowing the hydrophone sensitivity $H$, for
each frequency $f$, the projector voltage sensitivity $T$, can be calculated using the formula:

$$T = 20 \log \left( \frac{V_d}{V_p} \right) - H.$$

**Fig. 1:** The experimental setup and a photo of the projector and hydrophone.

![Experimental setup and photo](image1.png)

**Fig. 2:** The recorded signal (blue) and its part used to calculate the steady state voltage (red).

![Recorded signal and steady state voltage](image2.png)

The projector sensitivities measured for the V318 and V389 transducers are shown in Figs 3 and 4, respectively. The thick lines appearing in the graph, result from thinner merged lines, where each single line corresponds to a specific distance between projector and hydrophone. It can be seen that the sensitivity is not constant as the distance increases but it converges to a limit. The spread is larger for high frequencies and this fact is more apparent in the V389 (which has larger diameter than the V318). This phenomenon can be explained by the proximity criteria imposed in such measurements. Indeed the proximity criterion for a projector comes from the requirement that the pressure must be that of a spherically diverging wave, [1]. An approximate criterion for a uniform circular piston is,

$$d \geq \frac{\pi a^2}{\lambda}$$

where $a$ is the piston radius and $\lambda$ is the wavelength. It is apparent that the proximity criterion is satisfied for both transducers at a distance of 50 cm.
Fig. 3: Sensitivity chart for transducer V318

Fig. 4: Sensitivity chart for transducer V389
3. PROJECTOR CALIBRATION (SHORT PULSES METHOD)

We will now examine if the short pulses method which is discussed in [2], can be used for calibration purposes. Results only for the V389 transducer will be presented in this work. For this purpose we used the same experimental configuration, shown in Fig. 1, as in the previous case. Here only five cycles of a sinusoidal wave train signal with a specified frequency were used.

The distance between source and receiver was set to 25 cm while the frequencies spanned a range from 300 kHz to 700 kHz with a step of 40 kHz. The test signal used in this method was a pulse of Gaussian shape. This test signal along with its spectrum is shown in Fig. 5a. Driving the source with this as input results in the signal shown in Fig. 5b, (respective signal spectra appearing on the right column). Dividing the spectrum of Fig. 5b by the one of Fig. 5a (in a range up to 1 MHz), we obtain the transducer transfer function \( H(f) \), shown in Fig. 6.

![Fig 5: a) The Gaussian input signal with its spectrum and b) The signal received at the hydrophone with its spectrum.](image1)

![Fig. 6: The transfer function \( H(f) \)](image2)
Assuming that a five cycle signal has to be produced at the receiver, the procedure shown in figure 7 is applied for each frequency. The resulting signals (lower left) were used to drive the projector. The signals recorded were the well defined 5 cycle waveforms at the specified frequency (shown in red at Fig. 10b). Measuring the amplitude of the received signal and using the hydrophone’s sensitivity the results, shown in Fig. 8 as red circles, were obtained. Comparing these data with the the curve obtained with the first method (blue line, Fig.8) we notice a big discrepancy between the corresponding values. In order to alleviate this deviation the following procedure was used.

**Fig 7: The procedure for the creation of the input signal**

**Fig. 8: The sensitivity of the V389 as measured by the typical method (blue line) and its sensitivity using the second method (red circles)**
In Fig. 9 the prescribed input signal spectrum corresponding to a 5 cycle wave train at 500 kHz is shown. It can be seen that in addition to the 500 kHz frequency component, which is apparent in this graph, there are other frequencies with large amplitude present. This means that the energy of the signal sent, is spreading in a broader frequency range and thus the energy for frequencies around 500 kHz is less than expected. To compensate for this we removed all frequencies shown in Fig. 9 except for the narrow band around the 500 kHz (shown in red at Fig. 9). Applying the Inverse Fourier transform in this signal we obtain a different input signal depicted in Fig 10a. Driving the source with this signal we obtain the signal at the hydrophone’s position, shown in blue in Fig 10b, while the signal obtained using the original input signal (Fig. 7d) is shown as red. If we apply this technique for all predetermined frequencies and measure the amplitudes of the received signals at each frequency for the calculation of the sensitivity, we obtain the results shown as red circles in Fig. 11. We can now observe that the results are quite similar to those obtained with the typical calibration method.

**CONCLUSIONS**

The calibration of two ultrasound projectors using the traditional method was performed in the first part of this work, while a new method, appropriate for low frequencies in medium sized tanks, was presented in the second part. The latter is based on the short pulses generation method, [2]. When applying each single method, the calibration results were compared between them for one of the transducers. It seems that the direct application of the second method results in discrepancies when compared to the established method. Nevertheless, after appropriately modifying the input signal, results compare well with the established (traditional) method.
Although more experiments are needed in order to validate this methodology, the new proposed method appears to be the only approach for transducer calibration at low frequencies in medium sized water tanks.

![Fig. 11: The sensitivity of the V389 as measured by the typical method (blue line) and its sensitivity using the modified second method (red circles)](image)

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REFERENCES

DE-COHERENCE EFFECTS IN UNDERWATER ACOUSTICS: SCALED EXPERIMENTS.

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Abstract: We reproduce, using scaled experiments in a water tank, the effects of scattering phenomena responsible for the degradations of sonar system performances in oceanic environment (typically, the small sound speed fluctuations associated with linear internal waves). We reproduce a wide panel of scattering effects, spanning from “simple” phase aberrations up to radical changes in the sound field structure (appearance of caustics). An experimental protocol was developed. It consists in transmitting a high-frequency wave train (ultrasonic pressure field around 2MHz) through wax lenses with randomly rough faces, that induce distortions comparable to those that would be observed at sea at around 1kHz in the case of a lower frequency acoustic signal travelling through a linear internal wave field. Using a 3-D printer, we were able to manufacture lenses with a randomly rough face characterized by its amplitude and vertical and horizontal correlation lengths. The dependence of the various parameters involved in the experiment (related to the object, distance of propagation, frequency, ...) were studied using simulation programs allowing to measure the average number of eigen rays and the phase difference between the extreme micro paths. Those two quantities are useful to compare our results to what was obtained in the literature, in particular to Flatté’s dimensionless analysis. The propagation through the lenses was then studied in a water tank using virtual arrays (automatic displacements of a hydrophone). We represent the results using the acoustic envelop in order to observe wave front distortions or appearance of caustics. Measurements of the coherence function and, hence, of the radius of coherence, are carried out. Finally, we observe degradation of the performances of a localization algorithm.

Keywords: De-coherence, Tank Experiments, Fluctuations, Dimensionless Analysis.
1. INTRODUCTION.

We focus here on the topic of wave propagation in random media (WPRM). Even though a considerable amount of contributions to the field is available in the literature ([1-5]), our thought is that providing experimental data acquired in controlled environment would be of great help in order to understand the involved physical phenomena. Our main objective is therefore to develop an experimental protocol allowing us to measure in a water tank (i.e. at reduced scale) signal distortions comparable to what would be observed in the case of a lower frequency sound wave traveling through a spatially fluctuating ocean. As an example, linear internal waves (LIW) are responsible for perturbations in the underwater sound propagation, and induce some degradation of the array performances [6-9].

The objective of this research, in fine, is to provide some corrective signal processing techniques in order to compensate for these de-coherence effects [10].

2. SIMULATION STUDY.

In order to anticipate for the induced distortion of the acoustic signals, we developed a ray tracing program allowing us to calculate the average number of eigen rays \( \langle N_{eig} \rangle \) and the rms phase difference between the extreme micropaths \( \Delta \Phi_{\text{rms}} \). According to Flatté [5], these two quantities are related to the dimensionless parameters \( \Lambda \) (diffraction parameter) and \( \Phi \) (strength parameter) used to classify the signal fluctuations. The relationships between the quantities previously cited have been verified in [11]. Hence, tracing rays through a specific wax lens allows us to anticipate for the fluctuation regimes involved in a given experimental configuration, as depicted by Fig.1.

![Fig.1: (a): Ray Trace – Vertical Direction – Wax Lens; (b): \( \Lambda \Phi \) Plane – 256 Sensors Vertical Array – \( f=2.25\text{MHz} \).](image)

3. EXPERIMENTAL CONFIGURATION.
Our idea is here to develop an experimental configuration that would allow us to observe acoustic signals obtained in the various regimes of saturation defined by Flatté. Thus, the goal of this study is to be able to observe perturbations of the acoustic signals similar to the ones observed in the case of a sound wave propagating through an internal wave field. This translates by distortions and folding of the acoustic wave fronts, and by the presence caustics in the measured pressure field.

The method we adopted for reproducing such phenomena in acoustic tanks is based on the propagation of an ultrasonic signal (f=2.25MHz) through an acoustic lens, and the measurement of the acoustic pressure field propagating through this object and throughout specific regions of the three-dimensional space. A diagram of the experimental configuration is given in Fig.2:

![Fig.2: Experimental Configuration Diagram.](image)

The physical properties of the material composing the object are important since they govern the way acoustic rays will be refracted. The material chosen here is referred to as *Machinable Blue Wax* (used in [12]). Its properties are listed in Table 1:

<table>
<thead>
<tr>
<th>Property</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Density</td>
<td>0.98</td>
</tr>
<tr>
<td>Longitudinal Wave Sound Speed [m/s]</td>
<td>1975</td>
</tr>
<tr>
<td>Shear Wave Sound Speed [m/s]</td>
<td>772</td>
</tr>
<tr>
<td>Longitudinal Wave Attenuation [dB/cm]</td>
<td>13 @ 2.25 MHz</td>
</tr>
</tbody>
</table>

*Table 1: Physical Properties - Machinable Blue Wax.*

It is mainly interesting to notice that the density of the material is very close to the one of water, meaning we can consider the density discontinuity to be quite negligible. In order for the experimental configuration to match the simulation results, the manufactured distorting object must be as close as possible from the one used in the simulation framework (input plane face and randomly rough output face with
$L_x = 3\text{mm}, \ L_y = 30\text{mm}$ and $\delta_x = 3\text{mm}$ . To do so, we developed a process leading to a sample realized in the appropriate material: first, the profile defined in simulations is interpolated and edited using a CAD software. The output file of the CAD software is then sent to a 3D printer that will produce a first version of the sample. Nevertheless, the material used by the 3D printer does not feature the appropriate acoustical properties (due to its honeycomb structure), meaning that the printed object will be used as a primary mold to produce the final distorting object in another material. Then, we used a molding silicone (RTV 2-RTV 123) in order to obtain a "negative" mold of the original profile. This silicone has the advantage to keep its shape at high temperature, and therefore, it allows us to pour the melted Machinable Blue Wax. This particular step of the manufacturing process was realized under the dome of an air pumping system in order to avoid the appearance of air bubbles at the surface of the sample. Fig.3 displays the sample at the different steps of the manufacturing process.

![Fig.3: Wax Lens Manufacturing Steps: (a) CAD Software Output; (b) Actual Manufacturing Steps Outputs.](image)

The experimental measurements are conducted in a 3m long, 1.5m wide and 1m deep water. It is filled with fresh water that is controlled using a temperature probe. The acoustic equipment (transducer and hydrophone) as well as the distorting lens are fixed on motorized rails that are driven by a computer interface [13]. The transmitted signal is generated by a random function generator. The signal is then sent through the distorting lens in water using a Panametrics transducer V306-SU, centered at 2.25MHz. The distorted acoustic pressure field is then recorded using a Precision Acoustics Needle Hydrophone. The recording of the sound pressure field is completed at various positions of the hydrophones. The program allowing to control the displacements of the motorized rails also commands automatic interpolated positions, given an initial and a final coordinates and a step size.

4. EXPERIMENTAL RESULTS.

In order to obtain different statistical realizations of the experiment, the depth of the source was changed, so that different regions of the random profiles were acoustically highlighted.
4.1 WAVEFRONT DISTORTIONS.

Fig. 4 shows the acoustic envelop of the received signal. This quantity is calculated using the Hilbert transform of the raw received wave train:

$$A_j(t) = \log_{10} \left( p_j(t) + i H \{ p_j(t) \} \right),$$

(1)

where $p_j(t)$ denotes the acoustic pressure recorded at receiver $j$ and $H \{ \cdot \}$ is the Hilbert transform. Distortions are observed when we compare the signal propagated through the wax lens (here at the first distance and center source depth) with the signal propagated in water only. Especially, we notice shadow zones and caustics in the structure of the vertical array wavefront.

4.2 COHERENCE FUNCTION.

We also compared the coherence function calculated with the output of a simulation program presented in [11] with experimental results of the same configuration.

$$C(k, \epsilon) = \left\langle \tilde{p}_i(f_c) \tilde{p}^*_i(f_{c+k}) \right\rangle, \forall l \in [1, N], \forall k \in [1, N-1].$$

(2)
Where $k$ is the spacing index, $\varepsilon$ represents the spacing between two consecutive sensors, $\tilde{p}_l$ stands for the Fourier transform of the pressure recorded at receiver $l$, $f_c$ is the signal center frequency and $\cdot^*$ denotes the complex conjugate.

Fig. 5: Average Coherence - (a): VA; (b) HA.

Fig. 5 displays the average coherence function along the linear array for the first propagation distance ($d_{\text{src/rcvr}} = 0.23 \, \text{m}$). The agreement between the two cases is satisfying for the main lobe, and therefore for the radius of coherence, whose value can be related to the array gain [7]. We observe here a strong degradation of the array performance. Even if the vertical array case displays a decrease of the radius of coherence, the influence of the transducer directivity is noticeable: in the unperturbed medium case, the coherence function is very close to the horizontal array case.

4.3 DETECTION ALGORITHM.

In order to measure the influence of the propagation of acoustic signal through a perturbed medium, we also developed a near-field and range-dependent beam forming routine, i.e. a focalization algorithm.

$$b(x, y) = 20 \log \left[ \frac{1}{N} \sum_{l=1}^{N} w_l p_l \left( t - \frac{\|MS_l\|}{c} \right) \right], \quad (3)$$

where $w_l$ is the weight vector corresponding to sensor $l$, $S_l$ is the $l$th sensor, $M$ is a point of coordinates $(x, y)$ and $c$ is the sound speed in water.

The localization of the source in the case of propagation through an unperturbed medium (Fig. 5 (a) and (b)) corresponds to the maximum of the represented function. The position of the source is here obtained with a good precision and matches the experimental configuration. As depicted in Fig. 5 (c) and (d), the localization of the source is made more difficult when the focalization algorithm is applied to signals propagated through the wax sample and measured.

If we focus on the vertical antennae case, we observe strong distortions of the focalization output.
5. CONCLUSION

In this paper, we presented an experimental protocol allowing us to measure distortions of acoustic signals propagated through a wax lens that we also manufactured. We conducted series of experiment in a controlled environment leading to degradation of linear array performances. We anticipated for the experimental results with ray tracing programs allowing us to classify the signal fluctuations in Flatté's $\Lambda\Phi$ plane. We observed distortions of the signal envelop both for vertical and horizontal linear array and measured the degradation of their detection performances with the calculation of the coherence function. The output of a spherical beam forming algorithm confirmed the fact that the distortion effects prevent the algorithm from accurately detecting the position and direction of the source. Future work will consist in improving the experimental protocol in order to quantitatively scale the distorted acoustic field characteristics (such as the correlation length to wavelength ratio) to the ocean case. Also, signal processing techniques including de-coherence model based optimal filter or beacon based techniques will be tested on both synthetic and experimental data.

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REFERENCES

ACOUSTIC ECHO REDUCTION AND INSERTION LOSS OF TILES

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Abstract: A method is presented to determine, in water or air, the low-frequency acoustic Echo Reduction (ER) and Insertion Loss (IL) of infinite flat tiles by using small finite tiles. A signal generator, power amplifier, and a transducer are used to radiate acoustic waves that are incident on the finite panel. The pressures in the incident, reflected, and transmitted waves are measured using a hydrophone. A signal is designed such that the incident pressure is of short duration and rich in low-frequency content. Hydrophones are positioned close to the tile on either side but just sufficiently far away to prevent interference between the reflected and transmitted waves and the incident wave. The small tile is just sufficiently large enough to prevent interference between the reflected and transmitted waves and the wave scattered by the edge of the panel. Measurements on steel panel are used to show that the method is accurate even at ka = 1 where k is the acoustic wavenumber and a is the side of a square tile.

Keywords: Echo reduction, Insertion loss, small-area tile, finite-size panel
1. INTRODUCTION

Echo Reduction (ER) and Insertion Loss (IL) [1] achieved by covering a large, essentially infinite, area with acoustic tiles are of interest. The tiles are often made in large numbers but in relatively small lateral sizes. Further, for test purposes, it is usually convenient and often necessary to use small area tiles. Particularly, in underwater applications, where the performance of the tile under hydrostatic pressure is of interest, the tank in which the measurements are done and the tile are small with respect to the wavelength in water. In this paper, a method is presented to determine the ER and IL of infinite area flat tiles by using small tiles.

The frequency-dependent ER and IL are functions of the elastic properties of the materials used in the tiles, the thickness of the tile, the shape and size of cavities in the tile, the angle of incidence, the backing plate, hydrostatic pressure, temperature, etc. However, they should not depend on the method used to measure them or the lateral size of the tile. The target strength of a tile of finite size is dependent on the lateral size of the tile [2].


In the method presented in this paper, a signal is designed to drive the power-amplifier such that the pressure radiated by the projector, measured using a hydrophone at some distance from the projector and close to the panel, is of short duration and is rich in low-frequency content. Papadakis et al. [7] present a method to obtain a few cycles of cw. Jing and Fung [8] present a method to design a signal such that reflections from the end of a pipe are cancelled. Here, time-windows are used to select only the short-duration signal and thus increase the Signal to Noise Ratio at low frequency. The measured ER and IL of a steel plate with $ka$>1 are in good agreement with values computed for an infinite steel plate.

2. EXPERIMENTAL SETUP

A schematic of the experimental setup is shown in Fig. 1. The tile, a projector on one side to generate acoustic waves, a hydrophone on the same side of the tile as the projector to measure the incident and reflected waves, a hydrophone on the other side of the tile to measure the transmitted wave are shown.

A signal generated using Matlab is fed to the power amplifier (Model L10, M/s. Instruments Inc., USA) through a multi-function data acquisition system, MF DAQ (Model NI 6115, M/s. National Instruments, USA). The amplified signal from the power amplifier is then fed to the projector (Model B & K 8104, M/s. Bruel & Kjaer, Denmark or Model NP 20, M/s. NPOL, India) that radiates acoustic pressure. The acoustic pressure wave is sensed by the hydrophone (Model B&K 8104, M/s. Bruel & Kjaer, Denmark). The signal is then amplified and conditioned using the pre-amplifier (Model B& K Nexus, M/s. Bruel & Kjaer, Denmark). The conditioned signal is connected to the MFDAQ for further analysis.
When a plane wave is incident on an infinite tile with the same fluid on either side, ER and IL are defined as follows:

\[ ER(\omega) = 20 \log_{10} \frac{|A_i(\omega)|}{|A_r(\omega)|} \]  

(1)

and

\[ IL(\omega) = 20 \log_{10} \frac{|A_i(\omega)|}{|A_t(\omega)|} \]  

(2)

where \( A_i(\omega) \), \( A_r(\omega) \), and \( A_t(\omega) \) are the pressure amplitudes of the incident, reflected, and transmitted plane waves and \( \omega \) is the angular frequency.

3. TRANSFER FUNCTION

The transfer function of the transmit system is defined as

\[ H(\omega) = \frac{V_i(\omega)}{V_s(\omega)} \]  

(3)

where \( V_s(\omega) \) is the output voltage of the signal generator; \( V_i(\omega) = M(\omega) P_i(\omega) \) is the output voltage of the hydrophone in response to the incident pressure \( P_i(\omega) \), and \( M(\omega) \) is the free-field acoustic sensitivity of the hydrophone. \( H(\omega) \) can be determined by measuring \( V_s(\omega) \) and \( V_i(\omega) \) and finding the ratio. However, when noise is present, it is better to compute it by using

\[ H(\omega) = \frac{V_i(\omega)V_s^*(\omega)}{V_s(\omega)V_i^*(\omega)} = \frac{G_{XY}(\omega)}{G_{XX}(\omega)} \]  

(4a)

or

\[ H(\omega) = \frac{V_i(\omega)V_s^*(\omega)}{V_s(\omega)V_i^*(\omega)} = \frac{G_{YX}(\omega)}{G_{YY}(\omega)} \]  

(4b)

where \( G_{XX}(\omega) \), \( G_{YY}(\omega) \), and \( G_{XY}(\omega) \) are the auto spectra of the input and output, and the cross spectrum of the input and the output, respectively. Equations (3), (4a), and (4b) yield the same results when there is no noise.

In all the cases, the time-dependent voltages \( v_s(t) \) and \( v_i(t) \) at the input and output, respectively of the transmit system are measured where \( t \) denotes time. Then, a Fourier-transform is computed to determine the frequency-dependent voltages. Upper case and
lower case letters are used to denote frequency-dependent and time-dependent functions, respectively.

In principle, the transfer function can be determined by using any input signal that is not zero at the frequency of interest. It is better to measure the transfer function with the tile in place so that it represents the system at the time of measurement.

The responses of the hydrophone (on the same side of the tile as the projector) to the incident wave from the projector that directly reaches it and to the wave reflected by the tile are termed as the incident response and reflected response, respectively. Its responses to waves scattered from the edge of the tile and the boundary of the tank are termed as the edge and boundary response, respectively. The instants at which the incident, reflected, edge, and boundary responses start are $t_{i}\,^s$, $t_{r}\,^s$, $t_{e}\,^s$, and $t_{b}\,^s$, respectively. The instants at which the incident and reflected responses end are $t_{i}\,^e$ and $t_{r}\,^e$, respectively. The duration of the reflected response, $t_{r}\,^e - t_{r}\,^s$ is always greater than that of the incident response, $t_{i}\,^e - t_{i}\,^s$ because of the finite thickness of the tile. Without using a specially designed signal, the direct response usually does not end before the reflected response begins; and the reflected response does not end before the edge response or the boundary response begin; and it is not possible to separately measure the incident and reflected pressures and use them to compute the ER. Similar difficulties arise in the measurement of IL.

The spectrum, at low frequencies, is best measured using long time records with high signal to noise ratio. The measured transfer function of the transmit system, $H(\omega)$, is shown in Fig. 2.

![Fig. 2: Transfer function of the transmit system.](image)

4. DESIGNED INPUT SIGNAL

The input signal is designed by using the measured transfer function and considering the following conditions that the incident and reflected responses should satisfy:

$$t_{i}\,^e < t_{i}\,^s, t_{r}\,^e < t_{r}\,^s, \text{ and } t_{r}\,^e < t_{b}\,^s \quad (5a, 5b, 5c)$$

Now, consider the case where the tile is square of side $2a$ or a circle of diameter $2a$. Let the distance between the projector and the hydrophone used to measure ER be $2d$ and the distance between the hydrophone and the tile be ‘$e$’. Let the speed of sound in water or air be $c$ m/s. It follows that $t_{i}\,^f = 2d/c$, $t_{r}\,^f = 2(d + e)/c$, and Eq. (5a) is satisfied when
The edge response starts at \( t^e_e = \frac{\sqrt{[(2d + e)^2 + a^2]^{0.5} + [e^2 + a^2]^{0.5}}}{c} \) and this should be after the reflected response ends. The duration of the reflected response, \( t^r_e - t^r_s \), may be \( N \) times the duration of the incident response. Therefore, Eq. (5b) is satisfied if

\[
\frac{2(d+e)}{c} + \frac{NT_0}{2} < \frac{\sqrt{(2d+e)^2+a^2} + \sqrt{e^2+a^2}}{c}
\]  

(6b)

Next, consider a tank with a boundary at a distance \( g \) from the line joining the projector and the hydrophone. The boundary response starts at \( t^b_e = \frac{2(d^2 + g^2)^{0.5}}{c} \). Therefore, Eq. (5c) is satisfied if

\[
\frac{2(d+e)}{c} + \frac{NT_0}{2} < \frac{2 \sqrt{d^2+g^2}}{c}
\]  

(6c)

One incident response that can result in Eqs. (6a), (6b), and (6c) being satisfied is \( v_i(t) = A \sin(\omega_0 t) \) for \( 0 \leq t \leq T_0/2 \) where \( A \) is the amplitude of the response and \( T_0 \) is the period corresponding to \( \omega_0 \). From Eq. (6a), \( T_0/2 \) should be less than \( 2e/c \). For the special case of \( e = 0.1m \), it therefore follows that \( f_0 = \omega_0/(2\pi) > c/4e > 3750 \) Hz. Eqs. (6b) and (6c) are used to obtain other constrains on the duration of the direct response and hence the frequency \( f_0 \). The designed input signal, for this case, is

\[
V_s(\omega) = \frac{V_i(\omega)}{H(\omega)}
\]  

(7)

The corresponding time-domain signal is shown in Fig. 3.

The time-record showing the response of the hydrophone to the incident, reflected, edge, and boundary wave is shown in Fig. 4. It is seen that the first two are clearly separated from the others. Therefore, accurate ER and IL measurements are feasible.

5. ECHO REDUCTION AND INSERTION LOSS

The experimentally determined ER and IL of a 980 mm × 780 mm × 5 mm thick steel plate are compared with theoretical values for an infinite plate in Fig. 5 (a) and 5(b). At \( f > 300 \) Hz, the agreement is good and the coherence is > 0.8.
6. CONCLUSIONS

A method to accurately measure the Echo Reduction and Insertion Loss of effectively infinite tiles using small area tiles is presented. The measured values are in good agreement with theoretical values for a steel plate even at $ka = 1$. The method can be used to characterize tiles and panels in small tanks under controlled temperature and pressure conditions.

7. ACKNOWLEDGEMENTS

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REFERENCES

Session 24

Three-dimensional sound propagation models

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THREE-DIMENSIONAL RAY MODELLING OF HIGH-FREQUENCY UNDER-ICE SHALLOW-WATER SOUND PROPAGATION

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Abstract: An existing 3-D ray model, REV3D, is extended to allow modelling of high-frequency sound propagation in the presence of an ice cover. The extension includes computation of propagation loss and propagation time series, by incoherent as well as coherent summation of ray contributions. Ice thickness data are provided at the node points of a rectangular grid, with bilinear interpolation in between, and a rough ice sheet with keels can readily be modelled. So far, the upper ice interface is flat without ridges. An option is included to construct realizations of poorly known ice thickness grid data stochastically. Elliptically shaped ice floes without keels are first laid out randomly. Ice keels, shaped as ellipsoidal bowls with random thickness increments, are subsequently laid out between the floes. When a ray from a sound source in the water reaches the water-ice interface, it can be reflected back into the water or refracted into the ice as a compressional wave or as a shear wave. In order not to get a dramatic increase of rays to follow, a random choice can be made guided by the expected energy distribution among the different possibilities. Similar random choices are made when a ray reaches the upper ice interface or the ice-water interface from above. When the number of rays is increased, the total propagated energy is shown to converge to its correct value, but random contributions may remain in the coherently computed time series at a particular receiver. With a flat ice-water interface without keels, rays from the water are typically totally reflected. With ice keels, ice reflections may cause ray steepening, and rays are more easily transmitted into the ice, particularly as shear waves. As a result, additional energy losses are suffered. Computational results are compared to experimental 10 and 20 kHz propagation loss data from the Gulf of Bothnia, for a propagation range of 5 km.

Keywords: Kinematic and dynamic ray-tracing, selection of rays, stochastic modelling.
1. INTRODUCTION AND SUMMARY

The purpose here is to develop methods for studying high-frequency acoustic wave propagation in arctic areas, or in the Baltic Sea or Bothnian Sea, covered by ice. A three-dimensional (3-D) ray tracing model, REV3D, is extended to include effects from a rough ice cover with varying thickness. Effects of such boundaries are important at prediction of propagation loss of underwater sound [1], particularly at higher frequencies.

Rays are traced through the solid ice, and some relevant 3-D ray-tracing equations are included in Sec. 2. Theoretically, a ray from the water should be split when reaching the ice interface, with a reflected wave in the water and two waves transmitted into the ice: a compressional wave and a shear wave. To avoid a multiplicative increase of the number of rays to follow, however, the split is not made. Instead, a random choice can be made concerning which ray to follow and the ray amplitude is increased to avoid energy losses at the ice interaction. As shown in Secs. 2.3 and 2.4, such stochastic computations of propagation loss converge in the incoherent case to the correct value when the number of rays is increased, but random contributions may remain in the coherent case.

An effort is made in Sec. 3 to model 10 and 20 kHz propagation loss measurements in the Gulf of Bothnia involving an ice cover with basic thickness 0.65 m. Ice keels, whose exact location and size are not known, are included randomly according to a stochastic technique where keel-free floes are initially laid out. Modelling as well as measurement results indicate increased propagation loss because of the keels. Keel reflections often lead to ray steepening and additional bottom loss, which seems to be a more important loss mechanism than propagation through the ice. Effects of 3-D scattering in the azimuthal direction are small with the selected keel-distribution parameters.

2. THE REV3D PROPAGATION MODEL

REV3D is a 3-D ray model for computation of propagation loss, propagation time series, and reverberation in shallow-water environments. In its original version, [2] and [3], incoherent summation of ray contributions was used. More recently, versions with dynamic ray tracing, coherent ray summation, and Doppler effects have been developed, for applications to underwater acoustic communications [4] and active sonar with bistatic target returns and reverberation [5].

Some extensions have now been made to handle a possible ice sheet on top of the water column. The description in the present section focuses on these new features of the model.

2.1. Ray tracing in 3-D

A left-hand Cartesian $xyz$ coordinate system is introduced with horizontal coordinates $xy$ and depth $z$. The sound speed in the water is represented by range-independent profiles within horizontal $xy$ rectangles. In each such rectangle, the variation of the sound speed $c$ with depth $z$ is assumed to be “piece-wise $1/c^2$ linear.” Hence, each ray is a sequence of parabolic arcs. Bottom depths are given explicitly at the grid points for the horizontal rectangles, and bilinear interpolation is used in between. It follows that the intersections of a ray with the bottom can be calculated by solving second-degree algebraic equations.
An ice layer on top is introduced in a similar way, by also giving ice thickness data at the bottom-depth grid points. Again, bilinear interpolation is used in between, and the ray intersections with the ice layer can be obtained from second-degree algebraic equations.

Complex plane-wave reflection coefficients are computed for the interaction with the bottom. These reflection coefficients are computed for a bottom structure with plane fluid or solid layers locally following the bottom slope at the particular reflection point. Hence, the reflection coefficients become functions of frequency as well as incidence angle. No ray tracing is performed through the bottom. Ice reflections can be treated similarly.

The main option, however, is to trace rays explicitly through the ice. When a ray impinges on the ice from the water, the plane-wave water-ice reflection and transmission coefficients are calculated. In principle, there are three scattered waves: a reflected compressional wave into the water, and transmitted compressional and shear waves into the ice. A ray that has been transmitted into the ice layer may be reflected at the surface, and reflected as well as transmitted back into the water when subsequently meeting the ice-water interface. The reflections lead to compressional as well as shear waves. Only a homogeneous ice layer has been implemented so far. The ice layer should have a thickness of at least a few wavelengths for the ray approximation to be valid.

Some details on 2-D ray-tracing are given in [4] for the water column and in [6] for ice interaction. The following ray-tracing description focuses on 3-D interactions at an ice interface. The direction vector of a ray is written $\mathbf{e}_r$. With $|\phi| < \pi/2$, it can be expressed as

$$\mathbf{e}_r = (\cos \phi \cos \eta, \cos \phi \sin \eta, -\sin \phi)$$

in the introduced $xyz$ coordinate system. In particular, $\phi$ is the inclination angle, positive upwards, and $\eta$ gives the azimuth. The initial values for $x = (x, y, z)$, and $\phi, \eta$ at the source point are denoted $x_s = (x_s, y_s, z_s)$ and $\phi_s, \eta_s$, respectively.

### 2.1.1. Kinematic ray-tracing involving an ice interface

Upon reflection or refraction at the water-ice, ice surface, or ice-water interface, the direction vector $\mathbf{e}_r$ of the ray is changed according to Snell’s law to

$$\mathbf{e}_r = C_q \mathbf{e}_r + \Gamma \mathbf{e}_n = C_q [\mathbf{e}_r - (\mathbf{e}_r \cdot \mathbf{e}_n) \mathbf{e}_n] + H \mathbf{e}_n$$

where $\mathbf{e}_n$ is a unit normal of the interface at the interaction point, $C_q$ is the quotient of the pertinent sound velocities after and before the interaction, and underlining is used to denote the situation directly after the interaction. The constants $H$ and $\Gamma = H - C_q (\mathbf{e}_r \cdot \mathbf{e}_n)$ are obtained from $|\mathbf{e}_n| = 1$ and the condition that $H$ and $\mathbf{e}_r \cdot \mathbf{e}_n$ have the same (opposite) sign upon refraction (reflection). It follows readily that

$$H = \pm[1 - C_q^2 (1 - (\mathbf{e}_r \cdot \mathbf{e}_n)^2)]^{1/2}$$

The angles $\phi, \eta$ are obtained from the “underline version” of (1).

### 2.1.2. Dynamic ray-tracing involving an ice interface

A ray-centered left-hand Cartesian coordinate system with coordinates $r,u,v$ and unit vectors $\mathbf{e}_r, \mathbf{e}_u, \mathbf{e}_v$ are introduced along a given ray. These unit vectors are defined by (1) and

$$\mathbf{e}_u = (-\sin \phi \cos \eta, -\sin \phi \sin \eta, -\cos \phi), \quad \mathbf{e}_v = (-\sin \eta, \cos \eta, 0).$$
A point \((x,y,z)\) of reflection or refraction at an interface is changed to \((x+dx,y+dy,z+dz)\) for a nearby ray, as obtained by differential ray-tube changes \(d\varphi_s, d\eta_s\) to \(\varphi_s, \eta_s\). The corresponding ray direction vector \(e_r\) is changed to \(e_r + de_r\) with

\[
de_r = d\varphi \ e_u + \cos \varphi \ d\eta \ e_v.
\]  

(5)

In terms of the corresponding differential quantities \(du\) and \(dv\),

\[
(dx,dy,dz) = \kappa \ e_r + du \ e_u + dv \ e_v,
\]

(6)

where \(\kappa\) is determined by the condition that \((dx,dy,dz)\) is orthogonal to \(e_n\).

The differential changes \(d\varphi\) and \(d\eta\) of the inclination and azimuth angles \(\varphi\) and \(\eta\) upon interaction are also needed. It follows directly from the “underline version” of (5) that

\[
d\varphi = de_r \cdot e_u, \quad \cos \varphi \ d\eta = de_r \cdot e_v.
\]

(7)

Differentiation of (2) gives, with \(de_r\) obtained from (5),

\[
de_r = C_q \ de_r + \Gamma \ de_n + dC_q \ e_r + d\Gamma \ e_n.
\]

(8)

Second-order differentiation of the interface equation gives \(de_n = dx \ a + dy \ b\) for vectors \(a\) and \(b\). Since the sound velocities in each grid rectangle depend on \(z\) only, \(dC_q = (dC_q/dz)\) \(dz\). Finally, \(d\Gamma\) together with \(dH\) from (3) additionally involves \(d(e_r \cdot e_n) = de_r \cdot e_n + e_r \cdot de_n\).

The resulting expressions for \(d\varphi, d\eta\) show how a ray tube from the source is changed by ice interaction. Amplitude reductions by geometrical spreading can thus be computed.

### 2.2. Selection of rays

Rays are shot from the source point \(x_s\) with various initial inclination angles \(\varphi_s\) and azimuth angles \(\eta_s\). As noted above, two or three possibilities may exist for how a ray can proceed upon interaction with an ice interface. Evanescent reflected or transmitted rays may appear for certain angles, and such alternatives are discarded. Some additional selection procedure is needed for the propagating alternatives in order not to get a dramatic increase of rays to follow upon repeated ice interface interactions. Criteria for ray selection are discussed in [7],[8],[9], for example. The problem is facilitated in horizontally layered media, for which groups of kinematically and dynamically equivalent rays may be formed [7]. Starting with the laterally homogeneous case, Clarke [8] obtains a finite ray series by truncating expansions of reflectivity-method reverberation operators. Stabile et al. [9] develop a hierarchical technique to enumerate possible choices of subsequent ray segments, along with their phase types. Selection of the ones with which to proceed is based on amplitude estimates according to simplified reflection- and transmission-coefficients.

In the present paper, on the other hand, ray selection is made stochastically. When a ray reaches an ice interface, one and only one of the possible reflected and transmitted rays is selected, and its amplitude is magnified to keep the incident energy. The choice is made by randomization, governed by probabilities derived from the distribution among reflected and transmitted waves of energy flux normal to the ice interface at the contact point.

For waves with time \((t)\) dependence \(\exp(-i\omega t)\), complex notation for angular frequency \(\omega\), the time-averaged energy flux per unit area in the normal-vector direction \(n = (n)\) is

\[
(\omega/2) \mbox{Im}(\sigma_{ij} u_i^*) \ n_j,
\]

(9)
where $\mathbf{u} = (u_i)$ is the displacement vector, $\mathbf{\sigma} = (\sigma_{ij})$ is the stress tensor, the asterisk denotes complex conjugation, and the summation convention is used [10]. The energy flux is of course continuous across an interface. Coupling terms between pertinent incident, reflected, and transmitted waves appear in general when (9) is computed. For the particular case with elastic media (no absorption) on both sides of a plane interface, and all incident waves plane and propagating, the coupling terms disappear and the incident and scattered energy-flux sums for the individual participating plane waves are equal. This is related to the Hermitean character of a certain scattering matrix [11, Sec. 5.2.5].

Thus, for water and ice without absorption, the flux of an incident wave is partitioned among the scattered ones, and probabilities for the indicated random choice of ray continuation are readily obtained. At the frequencies considered (10-20 kHz), the water and ice absorptions are reasonably small in terms of dB per wavelength and the randomization is still applied.

### 2.3. Incoherent computation of propagation loss and time series

As in [12], all energy flow is confined to the rays, considered to be infinitesimally thin. Average intensities in REV3D are computed for receiver box volumes, obtained by a polar grid centred at the source in the horizontal plane and a division of the depth axis. Each ray contributes to the intensity average for a particular box according to the energy it carries per unit time and its arc-length within that box, divided by the box volume. Propagation loss (PL) curves are produced for the boxes in selected bearing directions from the source.

When the number of emitted rays from the source is increased towards infinity, the computed receiver-box intensities converge to the intensities that would be obtained if all ray-scattering alternatives at the ice interfaces were included. To realize this, consider $N$ rays with total energy $E$ emitted from a very small but otherwise arbitrary solid-angle patch. Each ray carries the energy $E/N$. Suppose that the possible path continuation alternatives to reach the receiver boxes, due to scattering and phase changes at the ice interfaces, involve a maximum of $L$ ice interaction points. At the $l$:th such point, the incident ray energy is distributed among a few alternatives (in some cases only one), indexed $i$, according to fractions $q_{il}$ such that $\sum_i q_{il} = 1$ for $l=1,2,..,L$. The particular ray-continuation alternative $i(1),i(2),..,i(L)$, with the choice $i(l)$ at $l$ for $l = 1,..,L$, carries the energy

$$\left(\frac{E}{N}\right) e[i(1),i(2),..,i(L)] q_{i(1)} q_{i(2)} .. q_{i(L)}$$

in each of $N$ rays to a selected receiver box. Here, the factors $e[i(1),i(2),..,i(L)]$, also depending on the selected receiver box, comprise the energy loss factors due to geometrical spreading and absorption.

With the random choice of ray continuation paths, on the other hand, alternative $i(1),i(2),..,i(L)$ appears with energy $(E/N) e[i(1),i(2),..,i(L)]$ for an expected number $N q_{i(1)} q_{i(2)} .. q_{i(L)}$ of rays. According to the law of large numbers, convergence to the correct energy sum $E \sum e[i(1),i(2),..,i(L)] q_{i(1)} q_{i(2)} .. q_{i(L)}$ appears when $N$ tends to infinity.

For a source pulse with a limited extent in time, it is not difficult to get an appreciation of how the propagated energy is distributed in time. Time traces for intensity averages in the different receiver boxes are obtained simply by positioning the energy from each ray at the appropriate time interval, with start time from the ray travel time and length given by the pulse length, along the time axis.
2.4. Coherent computation of propagation loss and time series

Each of the rays that are shot from the source at \(x_s\) is now considered to occupy a small initial ray-tube patch on the unit sphere. Except for absorption and interface scattering losses, the acoustic power of a ray is kept within its ray tube at propagation. This determines the change of intensity due to geometrical spreading, and the pressure \(p(x, t)\) at point \(x\) and time \(t\) can be expressed according to, cf. [5],

\[
[p(x) \ c(x)]^{-1/2} \ p(x, t) = \sum_{\alpha} \ [R(\alpha, x_s)]^{-1} \ [p(x_{\alpha}) \ c(x_{\alpha})]^{-1/2} \ p_{l}(t - T_{\alpha}) .
\]

Here, the sum on \(\alpha\) is for the rays with \(x\) within their ray tubes, and the corresponding traveltimes are denoted \(T_{\alpha}\). For a ray starting in a certain direction from \(x_s\), the corresponding ray-tube area at a point \(y\) along the ray is proportional to \([R(y, x_s)]^2\), where the positive function \(R(y, x_s)\), determined by dynamic ray-tracing as described in Sec. 2.1.2, is defined such that \(|y - x_s|^{-1} R(y, x_s) \approx 1\) for \(y\) close to \(x_s\). The density and the sound speed at position \(x = (x, y, z)\) are denoted \(\rho(x)\) and \(c(x)\), respectively. For points \(x\) close to \(x_s\), the pressure \(p(x, t)\) fulfills \(|x - x_s| \ p(x, t) \approx p_s(t)\), where \(p_s(t)\) denotes the source pulse.

Reflection and transmission coefficients at interfaces, typically computed at a centre frequency of the pulse, are readily included in (11). In particular, \(p_s(t)\) as well as its Hilbert transform \(p_s^*(t)\) are involved [4, Eq. (22)]. Propagation loss is derived from complex pressure amplitudes for the source “pulse” \(p_s(t) = \exp(-i\omega t)\) with angular frequency \(\omega\).

For the \(N\) rays emitted from a very small solid-angle patch, considered in Sec. 2.3, the particular ray-continuation alternative \(i_1(i_2) ... i(l)\), with the choice \(i_0\) at ice interaction point \(l\) for \(l = 1, ..., L\), now carries the energy given by (10) within each of \(N\) close ray tubes. With the random choice of ray continuation paths, on the other hand, alternative \(i_1(i_2) ... i(l)\) now appears with energy \((E/N) \ e[i_1(i_2) ... i(l)]\) within an expected number \(N \ q_{[i_1]} \ q_{[i_2]} ... q_{[i(l)]}\) of the close ray tubes. The summed energies for the \(N\) ray tubes apparently converges to its correct value when \(N\) tends to infinity. Convergence to the correct receiver time trace can not be expected for a particular receiver point, however, since it typically (in the absence of caustics etc.) acquires the energy \((E/N) e[i_1(i_2) ... i(l)]\) or zero instead of (10) from the considered ray-continuation alternative \(i_1(i_2) ... i(l)\).

Random contributions of ice interacting arrivals thus typically remain in the coherently computed time series. With ice keels that are not known in detail, a stochastic ingredient need not be unreasonable. Time traces for nearby receiver positions in an array can preferably be computed by time shifts for contributing ray components, cf. [5, Sec. 4.2].

3. ICE EXAMPLE

Homogeneous ice is assumed, with compressional and shear velocities 3700 and 1800 m/s, respectively. The corresponding absorption values are 0.4 and 1.0 dB per wavelength, respectively, and the ice density is 900 kg/m\(^3\). The sound velocity profile in the water is upward refracting, with an increase from 1405 m/s above the depth 43 m to 1422 m/s below the depth 58 m down to the bottom, which is at a depth of 70 m. Below the bottom, there is a postglacial clay layer (thickness 5 m, sound speed 1450 m/s, density 1400 m/kg\(^3\), absorption 0.2 dB/wavelength) above a half-space with glacial sand and fine grained moraine (sound speed 1550 m/s, density 1700 m/kg\(^3\), absorption 0.2 dB/wavelength).

Figure 1 shows 10 and 20 kHz propagation loss (PL) computational results, with source depth 15 m and receiver depth 5 m. Absorption in the water is included with 0.18 and 0.57
dB/wavelength at 10 and 20 kHz, respectively. The PL results are smoothed over a gliding window of length 250 m. For the flat case, the ice thickness is 0.65 m.

To model rough ice with keels, elliptically shaped ice floes with half-axes 30-100 m are first laid out randomly to cover 63.5% of the area. The keels are then laid out randomly between the floes to cover 4% of the area. They are shaped as half-ellipsoidal bowls, with major and minor horizontal half-axes 20-50 m and 2.5-5 m, respectively, and with a depth half-axis of 3 m. Together with the basic ice thickness 0.65 m and random thickness increments up to 3 m are, the ice depth within the keels varies between 0.65 and 6.65 m.

**Fig. 1:** The left and right panels show incoherent (Sec. 2.3) and coherent (Sec. 2.4) PL modelling results, respectively. In each panel, the black and grey curves are for 10 and 20 kHz, respectively. The upper two curves concern flat ice, while the ice keels are included for the four lower ones. Ice-keel results with 2-D modelling are shown dashed.

Corresponding experimental PL results, from the Gulf of Bothnia in April 2013, are only available for the range 5 km, with 64.0 dB for 10 kHz and 68.4 dB for 20 kHz. They agree reasonably well with the ice-keel modelling results in Fig. 1, and the larger PL at the higher frequency is of course due to increased absorption in the water and in the ice.

Ray-trace plots are shown in Fig. 2, with ice-keel reflections causing ray steepening and 3-D effects. The ice keels also facilitate transmission into the ice, mainly as shear waves, but the increased PL for the ice-keel curves in Fig. 1 is mainly caused by increased bottom loss induced by the ray steepening. Rays that penetrate the ice may already be bottom-interacting, and rays that are scattered out of the vertical plane with 3-D modelling may also lose energy with 2-D modelling, by ray steepening. In fact, isolation of the rays that penetrate the ice shows that they contribute very little to the pressure level, and PL computations without explicit ray tracing through the ice (using water-ice reflection coefficients only) turn out to provide similar results to those in Fig. 1.

**Fig. 2:** Ray-trace plots for a source at \((x,y,z) = (0,0,15)\) m. The left and right panels show six rays in the vertical \(xz\) plane and ten rays in the horizontal \(xy\) plane, respectively.

The time series in Fig. 3 (source depth 15 m, receiver depth 5 m) indicate an impulse response at range 5 km with significant multipath arrivals within a time interval of about 20 ms. This is in agreement with measurement results, but details of the relative
amplitudes among the different arrivals are hard to model. There are differences between the incoherent and coherent computations in the two panels, since different arrivals may interfere constructively or destructively in the coherent computations.

Fig. 3: 10 kHz incoherently (left panel, Sec. 2.3) and coherently (right panel, Sec. 2.4) computed impulse response time series at range 5 km. The pressure source impulse is 1 ms long. In the coherent case, showing absolute response values, it is a sine wave with unit amplitude at range 1 m and a Gaussian taper. In the incoherent case, showing square roots of intensity, unit intensity is assumed at range 1 m.

REFERENCES

NUMERICAL APPLICATIONS OF A HIGHER ORDER SQUARE-ROOT HELMHOLTZ OPERATOR SPLITTING METHOD ON MODELING THREE-DIMENSIONAL SOUND PROPAGATION

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Abstract: A higher order numerical algorithm has recently been proposed to split the square-root Helmholtz operator employed by the parabolic-equation (PE) method for modeling sound propagation. This operator splitting method includes multidimensional cross terms to yield a more accurate approximation, and, most importantly, it still permits efficient three-dimensional (3-D) PE numerical solvers, such as the Split-Step Fourier method and the Alternative Direction Implicit (ADI) Padé method. This paper will first review the numerical applications of this splitting method on forward models, and the importance of the cross terms in reducing approximation errors will be addressed. Numerical benchmark of a 3-D wedge problem will be shown in this proceeding paper to demonstrate the model performance, along with other computational examples presented in the talk on underwater sound propagation in the presence of internal waves, submarine canyons and seamounts. Development of 3-D tangent linear models and sound field sensitivity kernels will also be presented in the talk with examples considering complex environmental conditions. This square-root Helmholtz operator splitting method is readily applicable to modeling sound scattering from non-planar sea surface, and a numerical implementation employing the ADI Padé method is introduced. Lastly, suggestions for 3-D benchmark problems to test the accuracy and efficiency of numerical models are provided.

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Keywords: 3-D sound propagation, Parabolic-Equation Method, square-root Helmholtz operator
1. INTRODUCTION

The parabolic-equation (PE) approximation method, first introduced by Tappert [1] to underwater sound propagation modeling, has long been recognized as one of the most efficient and effective numerical methods to predict sound propagation in a complex environment. The advantage of this method is due to the fact that it converts/approximates the Helmholtz wave equation of elliptic type to a one-way wave equation of parabolic type. The conversion allows efficient marching solution algorithms (see Fig. 1) for solving the boundary value problem posed by the Helmholtz equation. This can reduce significantly the requirement for computational resources, especially for modeling three-dimensional (3-D) sound propagation.

To review the PE method, let’s first consider the following Helmholtz equation.

\[ \nabla^2 p + k_0^2 n^2 p = 0, \quad (1) \]

where \( \nabla^2 \) is the general 3-D Laplacian operator, and the Cartesian coordinate system \((x, y, z)\) is selected here for a uniform resolution. In Eq. (1), \( k_0 = \alpha/c_0 \) is the reference wavenumber, \( c_0 \) is the reference sound speed, and \( n \) is the index of refraction \( n = c_0/c = k/k_0 \). The index of refraction \( n \) is a spatial function and generally three-dimensional. When considering a medium with inhomogeneous density, one can use the density-reduced pressure variable approach [2] or the impedance reduced pressure variable approach [3] to include the density effects in the sound propagation model.

The PE approximation involves a factorization that makes the Helmholtz equation to be a transport equation with a convection operator of fractional (1/2) power, i.e., the one-way wave equation with the solution marching direction along \( x \):

\[ \frac{\partial}{\partial x} p = i k_0 \left( -1 + \sqrt{k_0^{-2} \nabla^2 + n^2} \right) p, \quad (2) \]
where a based line phase $k_0x$ has been removed from $p$, and $\nabla^2_{\perp}$ denotes the Laplacian operating on the transverse coordinates, $y$ and $z$. Note the one-way wave equation resulted from cylindrical coordinates will also have the same form [4]. To solve Eq. (2), the square-root Helmholtz operator,

$$\sqrt{k_0^2 \nabla^2_{\perp} + n^2} = \sqrt{1 + \varepsilon + \mu},$$

needs to be approximated for numerical computation. The operators $\varepsilon$ and $\mu$ are from $k_0^2 \nabla^2_{\perp} + n^2$, and different operator splitting algorithms are shown in the following to either (1) divide the Helmholtz operator into the free space propagator and the phase anomaly caused by sound speed gradients and waveguide boundaries, or (2) separate each one-dimensional spatial derivative. The first approximation to the square root Helmholtz operator, which leads to the standard narrow angle PE, was introduced by Tappert [5]:

$$Q_1 = 1 + \frac{1}{2} \varepsilon + \frac{1}{2} \mu.$$  \hspace{1cm} (4)

A wide angle PE with an improved approximation was proposed by Feit and Fleck [6], which has a twice better valid angle compared of the standard narrow angle PE:

$$Q_2 = -1 + \sqrt{1 + \varepsilon} + \sqrt{1 + \mu}.$$  \hspace{1cm} (5)

Note that both of these two PE approximations can be implemented with the Split-Step Fourier method [1] and numerical differential methods, such as Ref. [7].

A higher order splitting method has been recently proposed by Lin and Duda [8] to include multiple cross terms of $\varepsilon$ and $\mu$ in the approximation:

$$Q_3 = -1 + \sqrt{1 + \varepsilon} + \sqrt{1 + \mu} - \frac{1}{2} \left( -1 + \sqrt{1 + \varepsilon} \right) \left( -1 + \sqrt{1 + \mu} \right)$$

$$- \frac{1}{2} \left( -1 + \sqrt{1 + \mu} \right) \left( -1 + \sqrt{1 + \varepsilon} \right),$$

which is derived from the second-order Taylor series around $\sqrt{1 + \varepsilon} = 1$ and $\sqrt{1 + \mu} = 1$, and the non-commutative property of $\varepsilon$ and $\mu$ is kept, so $\varepsilon \mu \neq \mu \varepsilon$.

2. SPLIT-STEP SOLUTIONS

There are two approaches to split the square-root Helmholtz equation into either (a) two physical processes, i.e., the free space propagator and the phase anomaly, or (b) two sets of derivatives along each transverse direction. The operators $\varepsilon$ and $\mu$ obtained with the first approach are shown in the following.

$$\varepsilon = n^2 - 1 \quad \text{and} \quad \mu = k_0^2 \nabla^2_{\perp}.$$  \hspace{1cm} (7)
Note that the operator $\varepsilon$ implements the phase anomaly, and the operator $\mu$ constitutes the free propagation. Since the free propagator can be implemented by the Fourier transform, i.e.,

$$\exp\left[i k_0 \Delta x \left(1 + \sqrt{1 + \mu}\right)\right] p(x) = F^{-1}\left\{ \exp\left[i k_0 \Delta x \left(1 + \sqrt{1 - k_0^{-2} \mid k\mid^2}\right)\right] F\{p(x)\} \right\}, \quad (8)$$

this physical splitting method is also called the Split-Step Fourier (SSF) method [1]. The higher order SSF method has been introduced by Lin and Duda [8] using $Q_3$ shown in Eq. (6), and the solution is reviewed here:

$$p(x + \Delta x) = e^{\delta(1 + \sqrt{1 + \mu})/2} e^{\delta(-1 + \sqrt{1 + \varepsilon})} e^{-(\delta/2)(-1 + \sqrt{1 + \varepsilon})(1 + \sqrt{1 + \mu})} e^{\delta(-1 + \sqrt{1 + \mu})/2} p(x), \quad (9)$$

where $\delta = ik_0 \Delta x$, and the exponential operator with the cross terms can be implemented by the following iterative method.

$$e^{-(ik_0\Delta x/2)(NL+LN)} = 1 + \sum_m \frac{1}{m!} \left(-\frac{ik_0 \Delta x}{2} (NL + LN)\right)^m, \quad (10)$$

where the operators $N = -1 + \sqrt{1 + \varepsilon}$ and $L = -1 + \sqrt{1 + \mu}$.

Regarding the second splitting method of separating $\partial_y$ and $\partial_z$, the operators $\varepsilon$ and $\mu$ are shown in the following.

$$\varepsilon = w(n^2 - 1) + k_0^{-2} \frac{\partial^2}{\partial y^2} \quad \text{and} \quad \mu = (1 - w)(n^2 - 1) + k_0^{-2} \frac{\partial^2}{\partial z^2}, \quad (11)$$

where $w$ is a weighting factor between 0 and 1 to control the splitting of sound speed anomalies into $\varepsilon$ and $\mu$. The higher order PE solution determined by this splitting method has the general form as Eq. (9), but it cannot be implemented with the Fourier transform. Instead, we employ the following two Padé expansions to compute the solution with an Alternative Direction Implicit (ADI) method [9]:

$$e^{\delta(1 + \sqrt{1 + \mu})} p = \prod_i \frac{1 + \alpha_i g}{1 + \beta_i g} p, \quad (12)$$

and

$$\sqrt{1 + g} \ p = c_0 + \sum_i \frac{a_{ig}}{1 + b_{ig}} p. \quad (13)$$

The Padé expansion of exponentiated square-root operators Eq. (12) was introduced by Collins [7], and the rotated complex Padé expansion was derived by Milinazzo et al. [10].
3. ERROR ANALYSIS

Figure 2 presents the Split-Step Padé 3-D PE approximation errors associated with $Q_2$ and $Q_3$. As shown by Lin et al. [9], the approximation errors depend on two angles: one is the inclination angle $\theta$ steering away from $x$, and another angle is the orientation angle $\phi$ away from $y$. The errors are formulated as the following.

$$\varepsilon_2 = | -1 + \cos \theta - \varepsilon_y - \varepsilon_z |,$$

and

$$\varepsilon_3 = | -1 + \cos \theta - \varepsilon_y - \varepsilon_z + \varepsilon_y \varepsilon_z |,$$

where $\varepsilon_y = -1 + \sqrt{1 - \sin^2 \theta \cos^2 \phi}$ and $\varepsilon_z = -1 + \sqrt{1 - \sin^2 \theta \sin^2 \phi}$. These error terms are defined by $|\Delta \tau/\tau|$, where $\tau$ is the correct phase, and $\Delta \tau$ is the phase error. As shown in Fig. 2, when a wave vector $(\cos \theta, \sin \theta \cos \phi, \sin \theta \sin \phi)$ aligns with either $y$ or $z$, i.e., $\phi = 0$ or $\pi/2$, the phase error vanishes, yielding a perfect operator splitting. The maximum error occurs when $\phi = \pi/4$ (diagonal), and it bounds the overall approximation accuracy. Figure 2 also shows clearly that the higher-order operator splitting algorithm $Q_3$ performs better than $Q_2$, and the valid angle of $Q_3$ (measured by the inclination angle) is in general 10 deg larger than of $Q_2$ for any given error tolerance.

4. EXAMPLES

The first example is a classical 3-D wedge problem [11] with modified model parameters. The reference solution to check PE solutions is computed using Deane and Buckingham’s method of images [12]. The geometry of this wedge problem is shown in Fig. 3(a), and the model parameters are described in the following. The slope of the wedge
is 5 deg, the water sound speed is 1500 m/s, and the bottom sound speed is 1700 m/s. The medium absorption in water is neglected, but it is 0.5 dB/λ in bottom. Lastly, the water density is 1 g/cm³, and the bottom density is 1.5 g/cm³.

A 75-Hz point source is placed 2 km away from the apex at 100 m depth, and the SSF PE method is selected to compute the transmission loss (TL). The SSF PE method requires interface smoothing, and it has been shown that the higher order splitting algorithm allows sharper smoothing [8]. Here the smoothing width is set to be 1.25 m (1/16 λ). One can see in Fig. 3(b) that the regular wide-angle SSF PE using Q₂ cannot produce a solution matching the reference solution; on the other hand, the higher order SSF PE using Q₃ with cross terms yield a solution agreeing with the reference solution very well.

(a) Geometry of the 3-D wedge problem

(b) 75 Hz TL at 30 m depth

Fig.3: Comparison of higher order and regular SSF PE methods. (a) Geometry of the 3-D wedge problem; (b) TL solutions.

The second example is sound propagation in a waveguide with non-planar surface as shown in Fig. 4(a). The depth of the waveguide is 195 m, and the amplitude and wavelength of the idealized surface waves is 5 m and 1 km, respectively. The medium properties follows those in the wedge problem. A 25-Hz point source is placed 100 m below a surface wave crest, and the ADI Padé PE method with the higher order splitting algorithm Q₃ is utilized to compute the sound pressure field. The TL solution at 30 m depth is shown in Fig. 4(c), and one can clearly see the horizontal focusing along the surface wave crest that the source is placed below. This example demonstrates the capability of the higher order ADI Padé PE method on handling non-planar free surface conditions. A benchmark problem involving a semi-circular waveguide is developed to check the PE solutions with non-planar surface conditions, and the details will be presented in the talk.
Fig. 4: ADI Padé PE solution of 3-D sound propagation in an idealized underwater waveguide with wavy surface. The source is placed under a wave crest as shown in the panel (a), and the TL of 25 Hz sound at 30 m depth is depicted in the panel (b). Artificial sponge layers are placed on the sides to mimic a radiation boundary condition.

5. CONCLUSIONS

A higher order square-root operator splitting algorithm has been proposed for the parabolic-equation (PE) method to numerically model sound propagation in a waveguide. This splitting algorithm permits efficient numerical solvers, such as the Split-Step Fourier method and the Alternative Direction Implicit (ADI) Padé method, and it is readily applicable for 3-D problems. Two examples of an idealized wedge problem and a non-planar surface waveguide are presented in this paper, and other examples of internal waves, submarine canyons and seamounts will be shown in the talk.

As a conclusion remark, five different underwater sound propagation scenarios are suggested for 3-D benchmark problems: (1) sound propagation in a wedge/slope environment, (2) nonlinear internal wave waveguides, (3) submarine canyons, (4) seamounts and (5) 3-D waveguides with surface waves or roughness.

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REFERENCES


AN EXPLICIT ANALYTICAL SOLUTION FOR THE PROBLEM OF ADIABATIC SOUND PROPAGATION ALONG AN UNDERWATER CANYON WITH PENETRABLE BOTTOM

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Abstract: The problem of 3D sound propagation in a shallow-water waveguide featuring a canyon-type bottom inhomogeneity is considered. In the adiabatic case when the canyon depth is sufficiently small this problem may be solved using the mode parabolic equations. For the acoustic track directed along the canyon the analytical expressions for the mode amplitudes are obtained from these equations in terms of eigenfunctions determined by the shape of the canyon. This solution allows us to answer the question whether the sound will undergo usual cylindrical spreading or become partially trapped inside the canyon for a given shape of the latter. It may be also useful for the benchmarking of the 3D sound propagation models.

Keywords: Helmholtz equation, underwater canyon, normal modes, asymptotic methods
1. A WAVEGUIDE WITH CANYON-TYPE BOTTOM INHOMOGENEITY

Consider a shallow-water waveguide comprised by a water layer where the sound speed is \( c = c_w \) and density \( \rho = \rho_w \) overlaying the bottom halfspace where \( c = c_b \) and density \( \rho = \rho_b \). In a Cartesian coordinate system where \( z \) is depth and \( x, y \) are horizontal coordinates the bottom depth is described by a function \( h = h(x,y) \) (i.e. water-bottom interface is a surface \( z = h(x,y) \), water layer is the domain \( 0 < z < h(x,y) \)). In this paper we consider the sound propagation along an underwater canyon of the form

\[
h(x, y) = H_0 + \frac{D}{\cosh^2(\sigma y)},
\]

aligned along the \( x \) axis. Here \( H_0 \) stands for the “unperturbed” bottom depth, \( D \) is the depth of the canyon (i.e. its depression with respect to the \( z = H_0 \) level) and \( \sigma \) characterizes its width.

When the sound propagates along the canyon of the from (1) it may become trapped in the latter. This occurs due to the well-known (and experimentally confirmed) effect of the downslope refraction of acoustic waves in the ocean with the sloping bottom [1]. In our case it manifests in the form of the acoustic energy concentration near the \( x \) axis. Such ducting of the waves may prevent their cylindrical spreading (which typically makes the sound level to decrease as \( r^{-1/2} \) with the distance \( r \) from the source). In real ocean canyons are ubiquitous and, of course, their shape is not necessary similar to (1). However, for this specific shape of the canyon the 3D acoustic field \( p(x,y,z) \) produced by a simple CW point source may be computed analytically under certain assumptions (as is shown below). The most important of them consists in the adiabaticity of the propagation [1], i.e. we assume that the normal modes generated by the source experience only negligible coupling in the course of their propagation in our waveguide with underwater canyon (1). As is well-known, mode coupling may be neglected if the depth variations are sufficiently small, and the \( h(x,y) \) does not come close to the cut-off depth of the highest mode generated by the source. This implies a certain condition on \( D \) which depends on \( H_0 \), bottom properties and the sound frequency (see below).

The goal of present study is to obtain a closed-form analytical expression for acoustic pressure in the waveguide featuring the bottom inhomogeneity (1) near the canyon axis \( x \). The paper is organized as follows: in the Section 2 we outline the main asymptotic method (the MPE method [2,3]) for the solution of the adiabatic sound propagation problems, the Section 3 is devoted to the derivation of the expressions for mode amplitudes, and the Annexe A1 recalls the expressions for the mode functions of the Pekeris waveguide.

2. MODE PARABOLIC EQUATIONS

We use the so-called mode parabolic equations (MPEs) to describe adiabatic sound propagation in 3D waveguides [2,3]. MPEs accurately approximate adiabatic propagation of acoustical waves at small angles (in the \( xy \)-plane) to a given track aligned along the \( x \) axis. In this framework acoustic pressure is represented in the form of the modal decomposition
\[ p(x, y, z) = \sum_{j=1}^{N} A_j(x, y) \exp \left( \int_{0}^{x} k_j(x) dx \right) \phi_j(z, x), \] (2)

where \( k_j(x) \) and \( \phi_j(z, x) \) are the wavenumbers and mode functions satisfying the standard acoustical spectral problem at \( x=x, y=0 \), which is written as

\[
\begin{align*}
\gamma \phi_j \bigg|_{z=0} &= 0, \\
\phi_j \bigg|_{z=\infty} &= 0, \\
\gamma \frac{\partial \phi_j}{\partial z} \bigg|_{z=h-0} &= \gamma \frac{\partial \phi_j}{\partial z} \bigg|_{z=h+0} \\
\end{align*}
\] (3)

(here \( \omega \) denotes cyclic frequency of sound, \( c=c(z) \) and \( \gamma=\gamma(z)=1/\rho(z) \) are respectively the profiles of sound speed and inverse density at \( x=x, y=0 \)). \( A_j(x, y) \) are the modal amplitudes which obey [2,3] the following parabolic equations

\[
2ik_j A_{jx} + A_{jy} + ik_j A_j + V_j A_j = 0,
\] (4)

where potential \( V_j(x, y) \) is found by the formula

\[
V_j = \int_{0}^{\infty} \gamma \nu \phi_j^2 dz + h_1 \phi_j \left( \gamma \frac{\nu}{c^2} \bigg|_{z=h_0+0} - \gamma \frac{\nu}{c^2} \bigg|_{z=h_0-0} \right) + h_1 \gamma^2 \left( \frac{\nu}{c^2} \right)^2 \left( \rho \right) \bigg|_{z=h_0+0} - \rho \bigg|_{z=h_0-0} \right),
\] (5)

where \( h_0=h_0(x) \) is the mean (with respect to \( y \)) value of the bottom depth \( h(x, y) \), \( h_1(x, y) = h(x, y) - h_0(x) \) and \( \nu=2i\beta\omega^2/c^2 \) is a bottom attenuation factor: \( \beta \) is bottom attenuation in dB per wavelength and \( \eta=1/(40\pi \log_{10}(c)) \).

To compute acoustic pressure \( p(x, y, z) \) using (2) one usually solves acoustical spectral problem (3) at every point \( x \) (on some mesh in case of numerical solution), then the potentials \( V_j \) are computed from (5) and the mode parabolic equations (4) are solved for \( A_j \) (e.g. using finite differences). Finally the pressure field is assembled by (2).

In case of sound propagation in the underwater canyon (1) this solution procedure simplifies greatly (mainly because the spectral problem is the same for all \( x \)). Moreover, the solution of mode parabolic equations with the piecewise-linear potentials \( V_j \) corresponding to \( h_0=H_0=\text{const} \) and

\[
h_1(x, y) = \frac{D}{\cosh^2(\sigma y)},
\] (6)

may be obtained analytically in terms of eigenfunctions of the \( V_j \).
The solution of (4) in the domain $-\infty < y < \infty$, $0 \leq x < \infty$ requires Cauchy initial data $A_j(x=0,y)=A_{j0}(y)$ to be provided at $x=0$. We use the Gaussian initial condition [3] (it is shown that this condition optimally approximates the acoustic field produced by a point source in a Pekeris-type waveguide, cf. [3] for details):

$$A_j(0,y) = A_{j0}(y) = \frac{\varphi_j(z_s)\varphi_j(z_r)}{2\sqrt{\pi k_j\rho(z_s)}},$$

where $\varphi_j(z) = \frac{\gamma_{y_j}^2 c_j^2 - \gamma_{y}^2 c_r^2 + k_j^2 (\gamma_{y}^2 - \gamma_{y_j}^2)}{-\gamma_{y_j}^2 \varphi_j^2(h_0 - 0)(\rho_b - \rho_r)}$.

(7)

3. A SOLUTION OF THE MPE: EIGENFUNCTIONS OF POTENTIAL $V$

For the bottom relief $h(x,y)$ described by (1) the potentials $V_j$ in the mode parabolic equations (4) are of the form: $V_j = b_j h_1(y)$ where

$$b_j = \varphi_j^2(h_0) \left( \frac{\gamma_{y}^2 c_r^2 - \gamma_{y_j}^2 c_j^2 + k_j^2 (\gamma_{y}^2 - \gamma_{y_j}^2)}{-\gamma_{y_j}^2 \varphi_j^2(h_0 - 0)(\rho_b - \rho_r)} \right).$$

(8)

These equations may be therefore solved by the Fourier method (separation of variables). Substituting $A_j,\lambda(x,y) = \exp(i\lambda x/(2k_j))\psi_j,\lambda(y)$ into (4) we obtain a Sturm-Liouville problem for time-independent Schrödinger equation

$$\begin{cases}
\psi_{yy} + \left( \frac{D b_j}{\cosh^2(\sigma y)} - \lambda \right) \psi = 0, \\
\psi_{|y\to\infty} = 0, \\
\psi_{|y\to-\infty} = 0,
\end{cases}$$

(9)

where subscripts $j, \lambda$ are omitted for sake of clarity. Also the equation is complemented with the boundedness conditions at infinity in order to pick out bound state solutions (which correspond to the ducted waves). As is shown in ref. [4], the solution to the Schrödinger equation (9) writes as

$$\psi(y) = C \cosh^{-\kappa}(\sigma y) F(\kappa - s, \kappa + s + 1, \kappa + 1, (1 - \tanh(\sigma y))/2),$$

(10)

where $\kappa^2 = \lambda/\sigma^2$, $F(\alpha, \beta, \gamma, z)$ is the Gaussian hypergeometric function [5] and

$$s = \sqrt{1 + \frac{4D b_j}{\sigma^2} - 1\over 2}.$$ 

(11)

The boundedness conditions (9) at $y=\pm \infty$ require the hypergeometric function to turn into a polynomial [4,5]. This happens only if either its first or second argument is a negative integer or zero. Thus we obtain $\kappa - s = m$, where $m=0,1,2,\ldots$. From (9) and (10) it is
clear that $\kappa$ must be positive. We therefore conclude that there exist $M_j$ values of $\kappa$ satisfying

$$\kappa_m = s - m, \quad m = 1, 2, \ldots, M_j,$$

(11)

where $M_j < s < M_j + 1$. The corresponding values $\lambda_m = \sigma^2 \kappa m^2$ form the discrete spectrum of the Sturm-Liouville problem (9). Corresponding eigenfunctions may be expressed in terms of the Jacobi polynomials $P_m^{(a,b)}(x)[5]:$

$$\psi_{j,m}(y) = \frac{1}{Q_m} (\cosh(\sigma y))^{-\kappa_j} P_m^{(\kappa_j, s_j)}(\tanh(\sigma y)),$$

(12)

where $Q_m$ is a normalization constant such that

$$\int_{-\infty}^{\infty} \psi_{j,m}^2 dy = 1.$$

(13)

For the sake of completeness, let us recall that [5]

$$P_m^{(\kappa, \kappa)} = \frac{(-1)^m}{k! 2^k (1 - x^2)^{k + \kappa}} \frac{d^m}{dx^m} \left[(1 - x^2)^{\kappa + m} \right].$$

Now we can easily solve the Cauchy problem for (4) with the initial condition (7). Neglecting the continuous spectrum of (9) and retaining only the terms corresponding to the ducted waves (i.e. bound states), we obtain the following expression for $A_j(x,y)$:

$$A_j(x,y) = \sum_{m=1}^{M_j} \left[ \int_{-\infty}^{\infty} A_j^{(1)} \psi_{j,m}(y) dy \right] \exp \left( \frac{i\lambda_j}{2k_j} x \right) \psi_{j,m}.$$

(15)

Combining formulae (15) and (2) we obtain a closed-form asymptotic solution to the problem of sound propagation in a shallow-water waveguide (1) with an underwater canyon. This solution completely neglects mode interaction and therefore expected to be valid only when $D$ is small in comparison to $H_0$. More precisely, if for depth $H_0$ our waveguide admits $N$ trapped modes, then $H_0 + D$ must satisfy the following inequality:

$$H_{N+1} < H_0 + D < H_N,$$

where $H_N$ and $H_{N+1}$ are cut-off depth of $N$-th and $N+1$-th modes respectively. Note that we recalled the solution procedure for (3) and expressions for $\phi_j(z)$ necessary for the evaluation of $p(x,y,z)$ via (2) in the Annexe A1.
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ANNEXE

A1. SPECTRAL PROBLEM SOLUTION

For the sake of completeness let us recall the well-known formulae for the solution of spectral problem (3) in the two-layer waveguide. The discrete spectrum wavenumbers are determined from the equation for \( k \)

\[
\tan \left( \sqrt{\frac{\omega^2}{c_w^2} - k^2} H_0 \right) \rho_w \sqrt{k^2 - \frac{\omega^2}{c^2}} + \rho_b \sqrt{\frac{\omega^2}{c_w^2} - k^2} = 0, \tag{11}
\]

and the corresponding modes are of the form

\[
\varphi(z) = \begin{cases} 
\frac{1}{N} \sin \left( \sqrt{\frac{\omega^2}{c_w^2} - k^2} z \right), & \text{if } z \leq H_0; \\
\frac{1}{N} \sin \left( \sqrt{\frac{\omega^2}{c_w^2} - k^2 H_0} \right) \exp \left( \sqrt{k^2 - \frac{\omega^2}{c^2}} (H_0 - z) \right), & \text{if } z > H_0;
\end{cases} \tag{12}
\]
where $k$ satisfy (11) and $N$ is normalization constant:

$$
N^2 = \frac{H_0}{2\rho_w} - \frac{\sin\left(\frac{\omega^2}{c_w^2} - k^2 H_0\right)\cos\left(\frac{\omega^2}{c_w^2} - k^2 H_0\right)}{2\rho_w\sqrt{\frac{\omega^2}{c_w^2} - k^2}} + \frac{\sin^2\left(\frac{\omega^2}{c_w^2} - k^2 H_0\right)}{2\rho_b\sqrt{k^2 - \frac{\omega^2}{c_b^2}}}.
$$

(13)

Integral in the definition of $a$ in (8) may also be easily evaluated now:

$$
a = v_b \frac{\sin^2\left(\frac{\omega^2}{c_w^2} - k^2 H_0\right)}{2N^2 \rho_b \sqrt{k^2 - \frac{\omega^2}{c_b^2}}}.
$$

(14)
INCORPORATING A CROSS-MULTIPLIED TERM IN A THREE-DIMENSIONAL PARABOLIC EQUATION MODEL

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Abstract: Wide-angle parabolic equation (PE) based models are efficient and accurate tools for solving sound wave propagation problems in three-dimensional oceanic waveguides. For practical (or historical) reasons, most of them neglect cross-multiplied operator terms that appear naturally in the wide-angle square root operator approximation of the Helmholtz equation. It has been shown recently, both numerically and theoretically, that the use of a series of higher-order cross terms allows a reduction of the phase errors inherent to any PE computations and can thus handle greater propagation angles [1]. The cross terms were efficiently incorporated in a split-step Padé 3D PE algorithm. The objective of the present paper is to report some numerical results obtained incorporating a leading-order cross-term correction in an existing 3-D PE model, written in cylindrical co-ordinates, based on Padé approximations in both depth and azimuth, and a splitting operator technique. Various aspects of the numerical techniques used to handle the additional leading-order cross term are discussed. The improved accuracy of the now-fully wide-angle 3D PE model is assessed on benchmark numerical solutions.

Keywords: parabolic approximation, sound propagation modeling, 3-D effects.
1 Model description

We consider a multilayered waveguide composed of one water layer and one or several fluid sedimental layers. The geometry of each layer is fully three-dimensional. Cylindrical coordinates are used where $r$, $\theta$, and $z$, represent respectively the horizontal range, the azimuthal angle, and the depth (increasing downwards) below the ocean surface. Considering an harmonic point source of frequency $f$, located at $r = 0$ and $z = z_S > 0$, and assuming only outward propagation in range, the elliptic-type 3-D Helmholtz equation can be replaced by the following one-way equation

$$\frac{\partial \psi}{\partial r}(r, \theta, z) = i k_0 \left( \sqrt{I + \mathcal{X} + \mathcal{Y} - I} \right) \psi(r, \theta, z) \quad (1)$$

where $I$ is the identity operator, $\mathcal{X}$ is the 2-D depth operator in the $rz$-plane and $\mathcal{Y}$ is the azimuthal operator, defined as $\mathcal{X} := (n_\alpha^2 - 1)I + \frac{k_0^2 \rho}{\rho_0} \frac{\partial}{\partial z} \left( \frac{\rho}{\rho_0} \frac{\partial}{\partial z} \right)$ and $\mathcal{Y} := \frac{1}{(k_0 r)^2} \frac{\partial^2}{\partial \theta^2}$.

Here, $n_\alpha$ denotes the complex (to account for lossy layers) index of refraction, $k_0 := \omega/c_{\text{ref}}$ with $c_{\text{ref}}$ a reference sound speed, and $\rho$ the density, constant within each layer. The acoustic field $\psi$ is related to the acoustic pressure by $P(r, \theta, z) = H_0^{(1)}(k_0 r) \times \psi(r, \theta, z)$, where $H_0^{(1)}$ is the zeroth-order Hankel function of the first kind. The square-root operator present in Eq. (1) is then approximated as follows:

$$\sqrt{I + \mathcal{X} + \mathcal{Y}} \approx I + \sum_{k=1}^{n_p} \frac{a_{k,n_p} \mathcal{X}}{I + b_{k,n_p} \mathcal{X}} + \sum_{k=1}^{m_p} \frac{a'_{k,m_p} \mathcal{Y}}{I + b'_{k,m_p} \mathcal{Y}} - \frac{1}{4} \mathcal{X} \mathcal{Y}. \quad (2)$$

In a practical point of view, the two Padé series expansions appearing in the right hand side of Eq. (2) are very convenient since they allow for a very-wide angle propagation respectively in depth and in azimuth, the corresponding angular limitations depending on the two parameters $n_p$ and $m_p$ selected by the user. However, the now fully wide-angle capability of the 3-D PE model is attributed to the presence of the last (cross-multiplied) term in Eq. (2). Note that the cross-multiplied terms, including the leading-order cross term considered here, are neglected in most of existing 3-D PE models. Inserting the paraxial approximation given in Eq. (2) in the one-way 3-D equation (1), we obtain the following wide-angle 3-D PE model:

$$\frac{\partial \psi}{\partial r}(r, \theta, z) = i k_0 \left( \sum_{k=1}^{n_p} \frac{a_{k,n_p} \mathcal{X}}{I + b_{k,n_p} \mathcal{X}} + \sum_{k=1}^{m_p} \frac{a'_{k,m_p} \mathcal{Y}}{I + b'_{k,m_p} \mathcal{Y}} - \frac{1}{4} \mathcal{X} \mathcal{Y} \right) \psi(r, \theta, z). \quad (3)$$

The 3DWPE model [2] has been modified to handle the leading-order cross-multiplied term.

2 Numerical scheme

The wide-angle 3-D PE equation given in (3) is solved numerically using the following splitting method: Given the 3-D field $\psi$ at the discrete range $r_n$, $\psi$ is obtained at the next discrete range $r_{n+1}$ in three steps:
The first step consists in computing \( n_p \) intermediate fields denoted \( u^{(1)}(\theta, z) \), \( u^{(2)}(\theta, z) \), \ldots, \( u^{(n_p)}(\theta, z) \), by solving:

\[
[I + \mu^{(k)}_{-} A^{(n+\frac{1}{2})}] u^{(k)}(\theta, z) = [I + \mu^{(k)}_{+} A^{(n+\frac{1}{2})}] u^{(k-1)}(\theta, z), \quad 1 \leq k \leq n_p,
\]

where \( u^{(0)}(\theta, z) = \psi(r_n, \theta, z) \) and \( \mu^{(k)}_{\pm} := b_{k,n_p} \pm \frac{i \kappa_{0} \Delta r}{2} a_{k,n_p} \) with \( \Delta r \) denoting the increment in range. The discretization in depth is done using a piecewise-linear finite-element method. Denoting \( N \) and \( M \), respectively, the number of mesh points in depth and azimuth, at each range step, this first step requires the inversion for \( 1 \leq k \leq n_p \) of \( M \) algebraic linear systems of order \( N \) (with tridiagonal matrices), which corresponds to the calculation of the intermediate fields at successively adjacent azimuths \( \theta_1, \theta_2, \ldots, \theta_M \). Each set of linear equations is solved using a fast and robust direct (Gaussian) algorithm optimised for tridiagonal matrices.

The second step consists in computing \( m_p \) intermediate fields denoted \( v^{(1)}(\theta, z) \), \( v^{(2)}(\theta, z) \), \ldots, \( v^{(m_p)}(\theta, z) \), by solving:

\[
[I + \nu^{(k)}_{-} Y^{(n+\frac{1}{2})}] v^{(k)}(\theta, z) = [I + \nu^{(k)}_{+} Y^{(n+\frac{1}{2})}] v^{(k-1)}(\theta, z), \quad 1 \leq k \leq m_p,
\]

where \( v^{(0)}(\theta, z) = u^{(n_p)}(\theta, z) \), and \( \nu^{(k)}_{\pm} := b_{k,m_p} \pm \frac{i \kappa_{0} \Delta r}{2} a_{k,m_p} \). The discretization in azimuth is done using a higher-order centered finite difference (FD) scheme which allows a significant reduction of the required number of points in azimuth while still obtaining accurate solutions. At each range step, handling the azimuthal coupling terms requires the inversion for \( 1 \leq k \leq m_p \) of \( N \) algebraic linear systems of order \( M \), with entries in the upper right and lower left corners of the banded matrices to account for the continuity condition in azimuth. These inversions correspond to the calculation of the intermediate fields at fixed depth \( z_1, z_2, \ldots, z_N \). The bandwidth of the matrices depends on the order of the numerical azimuthal scheme. As in step 1, a direct (Gaussian) algorithm optimised for banded matrices is used.

The third step consists in computing \( u^{(1)}(\theta, z) \), by solving:

\[
[I + \frac{i \kappa_{0} \Delta r}{2} \times \frac{1}{4} (AY)^{(n+\frac{1}{2})}] u^{(1)}(\theta, z) = [I - \frac{i \kappa_{0} \Delta r}{2} \times \frac{1}{4} (AY)^{(n+\frac{1}{2})}] w^{(0)}(\theta, z),
\]

where \( w^{(0)}(\theta, z) = v^{(m_p)}(\theta, z) \) and \( \psi(r_{n+1}, \theta, z) = w^{(1)}(\theta, z) \). The discretization in both depth and azimuth is done using a second-order FD scheme which leads to a sparse, square matrix of order \( M \times N \), with a block-tridiagonal structure. All the blocks are tridiagonal matrices of order \( M \), with entries in the upper right and lower left corners. Note that the bandwidth being function of \( M \), the use of any direct algorithm as in step 1 and step 2, would require an excessive amount of memory storage (since storage must be allocated for the bandwidth in each row of the matrix), which would limit significantly the number of mesh...
points. Therefore, in order to effectively utilize the sparseness of the matrix, a non-stationary iterative algorithm equivalent to the preconditioned conjugate gradient iteration method for the normal equation is used at each step in range. For a 2-D or N×2-D computation, both steps 2 and 3 are ignored and \( \psi(r_{n+1}, \theta, z) = u^{(np)}(\theta, z) \). If step 3 is ignored, then \( \psi(r_{n+1}, \theta, z) = v^{(mp)}(\theta, z) \). It is important to notice that the convergence of the iterative algorithm used in step 3 deteriorates unreasonably when solving near \( r = 0 \). In order to bypass this problem, the computations are performed using a range-dependent number of points in azimuth [3]: at each step in range, the number of azimuthal points is selected such that the corresponding arclength increment \( \Delta s := r \Delta \theta \) remains less than a given fraction (to be defined by the user) of the acoustic wavelength \( \lambda \). In order not to deteriorate the quality of the discretization, the interpolation in azimuth is achieved at only specific discrete ranges. Handling the crossing term considerably slows down the computations. As will be seen in the next section, computations carried out ignoring step 3 allow a good description of the 3-D effects.

### 3 Numerical results

To assess the accuracy of the newly developed wide-angle 3-D PE model, we consider the now very-classical 3-D ASA wedge benchmark (three-dimensional extension of the original 2-D ASA wedge benchmark). An isotropic harmonic point source, emitting at 25 Hz, is placed at a depth of 100 m in an oceanic environment which consists of a lossless homogeneous water layer (sound speed: 1500 m/s, density: 1 g/cm\(^3\)) overlying a lossy half-space sediment bottom (sound speed: 1700 m/s, density: 1.5 g/cm\(^3\), and absorption: 0.5 dB/\( \lambda \)). No shear energy is assumed in the sediment. The wedge-like tilted water/sediment interface is described by the surface \( \{ z = h(r, \theta) \} \) where

\[
h(r, \theta) = 200 \left( 1 - \frac{r \cos \theta}{4000} \right).
\]

The water/sediment interface makes an angle of 2.86° with respect to the ocean surface at both \( \theta = 0^\circ \) (upslope direction) and \( \theta = 180^\circ \) (downslope direction) and is invariant along the \( \theta = 90^\circ \) and \( \theta = 270^\circ \) azimuthal directions. Note that the water depth at the source location is 200 m.

We compare in Fig. 1 the 2-D PE and 3-D PE solutions (plotted as black curves) with 2-D and 3-D reference solutions (plotted as gray curves). The 2-D reference solution is based on a normal mode expansion of the acoustic field. The 3-D reference solution is based on the image source method and was originally provided by Westwood (see for instance in Ref. [4]). We can observe that the 2-D PE solution, obtained with \( n_p = 2 \) and ignoring both step 2 and step 3 in the marching algorithm (see the discussion in the previous section), is in perfect agreement with the 2-D normal mode solution. The 3-D PE solution shown in the lower right subplot of Fig. 1 has been obtained considering step 1 with \( n_p = 2 \), step 2 with \( m_p = 2 \), and step 3 with \( \Delta s \leq \lambda/6 \). Also shown in the lower left subplot of Fig. 1 is the 3-D PE solution obtained ignoring step 3 in the computation. Vertical slices in the cross-slope direction of the transmission loss corresponding to the 2-D PE and 3-D PE solutions are also displayed in
Figure 1: Transmission loss (in dB re 1 m) curves at a receiver of 30 m in the cross-slope direction $\theta = 90^\circ$. The black curves correspond to the same 2-D PE solution (upper subplots) and to two distinct 3-D PE solutions (lower subplots). The gray curves correspond to reference solutions: 2-D normal mode solution in (a), same 3-D image solution in (b), (c), and (d). The 3-D PE solutions have been obtained computing (c) without step 3 or (d) with step 3.

Fig. 2. As expected, the 2-D solutions exhibit the interference pattern of the three (initially present) propagating modes for all ranges. The differences between the 2-D and 3-D solutions are very weak in the vicinity of the source, but become more and more pronounced as the propagation range increases. The 3-D effects have been explained in detail by several authors and correspond to intramodal interference effects, leading to the succession of three zones across-slope, with three propagating modes present in zone I, two propagating modes in zone II, and only one propagating mode interfering with itself in zone III. We observe the near perfect agreement between the (reference) 3-D image solution and the fully wide-angle 3-D PE solution that handles the leading-order cross term. We can also observe that the quality of the comparisons with the image solution deteriorates when the 3-D PE solution is computed ignoring step 3 (i.e. retaining only step 1 and step 2 in the calculations). We observe with great interest that the 3-D PE solution obtained without step 3 is able to reproduce the main 3-D effects (described above), though one can observe (mainly) a shift in the phasing of the solutions.
Figure 2: Transmission loss (vertical slices at constant azimuth, across-slope) corresponding to a 2-D PE computation (upper subplot) and 3-D PE computations without (middle subplot) and with (lower subplot) step 3. The 3-D solutions correspond to a Padé 2 expansion in both depth and azimuth ($n_p = 2 = m_p$).

References


Session 25

Towards Automatic Target Recognition: Detection, Classification and Modeling (of Underwater Targets)

Organizers: John Fawcett, Johannes Groen, Wolfgang Jans and Yan Pailhas
ITERATIVE TARGET RECOGNITION FOR PORT PROTECTION SYSTEM

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Abstract: Coastal areas’ protection is an important aspect in providing security for sea side establishments of high economic importance and/or military bases. The rise in underwater intrusion technologies, promotes terrorists to use unmanned vehicles in their attacks. A monitoring system approach discussed in this paper, considering the site investigation, using combining equipment types such as sonar, hydrophones, communication tools, multithreading software programs and so on, to get the maximum available data, which will be analyzed then sent to an appropriate processing part to detect any intruders’ threat. Design simulation was developed and tested. Hardware prototype was built with zero software versions, the trial result is shown. After trial, the current hardware-software version will be modified to produce a first version of our proposal system, which will correct some expected misleading results because of iterative method.

Keywords: Anti Intruder Systems, Port Protection, Shallow Water Noise
Introduction: Recent worldwide terrorist threats have generated a lot of interest in homeland security for many countries, which has led to increased operations in their local waters. These operations include defending the coastlines especially in shallow waters. Encountering underwater intruders is very insistent mission for any government; intruders include swimmers, divers, mini submarines, autonomous underwater vehicles, and more. Each class is considered as threat, which poses particular demands on surveillance, and hence, calls for different equipments and policies.

Site Investigation: International ports are diverse in their locations, surrounding environments, activities and regulatory regimes. Many are located in sensitive coastal and marine environments. To design an effective protection ant-intruder system, the site investigation is important step that includes infrastructure in electricity and communication, noise, surrounding barriers, auxiliary available cooperative institutions that can share its data, and so on.

Ambient Noise: Port activities have generated noise and vibration in both the terrestrial and marine environments, such as shipping, fauna, sonar systems, weapons testing, seismic tests, wind turbines.... Noise can be chronic, like the engine noise from a ship, or sporadic, like an occasional boom from seismic testing. Some sources are stationary, while some are mobile. Ambient noise in warm shallow waters level is alarmingly stronger (more than 25 dB higher) compared to the deep-water ambient noise. Propeller-driven ships have become the most dominant human-induced low frequency noise in the shallow water environment. Cavitations (the formation and then implosion of water vapour pockets which are caused by pressure changes across the propeller blade) has been identified as the main source of noise from moving vessels, figure 1[1,2] shows measured ambient noise spectra.

![Measured ambient noise spectra at wind speed 15kn.](image)

Monitoring System: The System was configured to enable monitoring on the screen that show real-time measuring and analyzing information of intruders for administrator through the screen. In design stage, there are some assumptions, which enable the generation of system requirements on which the design of the new system is based. The nature of the threat is in principle unknown and therefore remains the subject of continual debate and speculation among expert analysts and developers. Thus it imposes a significant, very pragmatic reduction
in the scope of a project, at least so far as the long list of sensor options is concerned. When designers and experts reach the same point of view, the system achievement starts, this system must include the main parts: detection, tracking, classifying, and prosecution. In our approach, we divide our design to hardware and intelligent parts, the hardware part include:

- **Hydrophone Array** is made up of four hydrophones placed in known locations. These hydrophones placed in a line on the seafloor, moored in a vertical line in the water column. Sound arriving at the array from a distant source, such as a submarine or intruder noise, will reach each hydrophone at slightly different times, depending on the direction from which the noise is coming. This time difference is known as the time-of-arrival-difference and can be turned into a direction. Using this information from all the hydrophones in the array, the noise direction can be calculated.

- **Side scan sonar** scanning sonar is located in specific depth and location produces a panoramic view of the surrounding area and can cover up to 360 degrees according to its location. Active type sonar will use with the principle of transmitting a sound signal created in a predetermined form with a predetermined frequency in fixed time frames and will then receive the echo from the underwater objects[3]. System processes the collected data while receiving, detects and starts tracking the potentially dangerous targets and rings alarm signals when necessary. The detected targets are presented graphically to the operator via a user interface.

- **Auxiliary gadgets** the design requires auxiliary instruments such as wind speed and direction, temperatures measurements devices, thermal cameras, lasers target designators and range finders, night sighting equipment, to have all necessary data which give system high processing efficiency. Also electronic components are required such as data loggers for each data source, filters and preamplifiers, batteries and so on. Some parts are fixed to its location and others install in buoy part, data may be transferred via radio link.

**Data Processing** Data collected from different sources, pass the main stages: thresholding, filtration, amplification, then processes while receiving, detects and starts tracking the potentially dangerous targets and alarm signals when necessary. The detected targets are presented to the operator via a user interface [4]. The suggested system is shown in figure 2. The detection and extraction process uses an adaptive threshold based on the noise statistics and additional data from auxiliary instruments, such as wind speed and temperature. The signal processing stage passes three main nested layers: thresholding, filtering and amplification which estimate: duration, frequency, time of arrival, SNR. Filtering and thresholding reduce the noise. The triangulation stage is embedded in processing stage; the output result uses to locate the noise source [5].

**Intelligent System Part:** the hardware part is followed by intelligent part, which includes software components supported by human operator.

**Firmware** Technology improvements in sonar systems over the past decade have focused on improvements in these key areas: hardware, tracking software and data fusion and systems integration. This part includes a list of tools such as computers, linkers, editors, test case generators, diagnostic checkers, library of functions, system test diagnostics and so on.
Figure 2: The suggested system structure

- **Software** includes system functional design, algorithms, and special developed software. The under development software is based on multithreading procedures to create high efficiency processing program, which can process the received data from multi channel data loggers parallel in reasonable time. The algorithm based on verification target simulator fixed on 800m from system, which produces signal similarly to intruder’s noise. During processing stages our developed software verifies the reference data presence. In case absence of this data, a readjustment filtering and thresholding stage generated, then repeat this strategy to reach reference signal detection. At this point the processing program raises succeed flag. In case reaching frame time (600 msec), the software omits the data frame result because of failing verification result. The developed software omits data frame before sending it to mosaic pseudo colour monitoring screen, the failure repetition raises full system alarm, which means the system is failed. Additional logical layer based on accumulative data frame processing. Verification the extracted data based to slow targets movement in water comparing to sound propagation, in the monitored area (about 800m) less than 600msec, target in water doesn’t move rapidly, so the change in signal is smooth that helps us to compare between data frames to verify the received data, and omits the non logical frames. We faced this case in first two trial days, then redeveloping procedures and criteria were done, after that this case were faced rarely, with trials we rebuild subcomponents and change design criteria to reach minimum system failure. So applying more than verification techniques (software, hardware, logical analyze judgement...) we reach maximum data processing efficiency.

- **Expert operator** who cans emphasis the system threat detection capability and decrease the false alarm ratio, the operator assets some machine decision like signature, which refers to characteristic markings, such as the auditory signature of a submarine, detected on sonar. The signals or waves received
could be typed (i.e. related to specific targets) for identifying characteristics. Skilled and experienced operators could provide reasonably accurate estimates of range, bearing, and relative motion of targets.

**Threat Analyzes:** After previous stages, the result analyzes then gives mixed human-machine decision; the final response may call for unsustainable readiness for instance. It makes the security system more immediate threat response of attack. Threat analyzes pass the main steps:

- Auto detect, track, classify, and alert at ranges sufficient to respond and intercept,
- Target localization and confirmation,
- Scale response according to defined rules depending with location and situation,
- Notify anti-intruder security equipage to take their role.

Threat analysis has proven itself many times, especially for military development. The mode of attack cannot be predicted with former certainties except to say that it will, until its final moments, probably blend in with legitimate civilian activity in the monitored area. The degree of uncertainty now about the threat may be among the more important transformations in the military. It may also be among the more longstanding distinctions between military combat and civilian security, with implications for port protection. Threat analysis motivates new technology, the asymmetric threat, because of its uncertainty, fails to impose the former unity and clarity on the design process—on the long series of design decisions, tradeoffs and exclusions, which must inevitably be made at many points throughout a design [6].

**Simulation** with technology achievements, the simulation play important role to reduce time and effort. Using simulation environments, like Matlab for signal processing and mathematic calculation, and Proteus for hardware including DSP techniques. We developed a first version of our proposal, that version now tested, these extracted data contains known shipping noise, swimmers, rain fail, and wind turbulence signals. The Proteus simulation thresholds the input data, then filters it then amplifies the received data. In Matlab simulation program, the signal classified and processed, then analyzes result to compare the detected object with known object, then assets the result, after that modifies the design and repeats the previous steps. We reached about fifty per cent of convergence between required and achieved result, that is very satisfy result in designing such systems for shallow water, that means a reasonable detection and satisfied false alarm ratio. By adding reference inserted signal the final detection results were enhanced, and false alarm ration was reduced to thirty per cent.

**Trial Results** The suggested system zero prototype version was done, in trial step we inserted a signal source with known parameters, after each frame reception, the processing program compares the current frame with previous one, this strategy reduces noise. Verification step based on detecting the reference signal, considering the recognition of inserted reference signal, readjustment filters and threshold process iteratively to reach reasonable convergence results between expected and real ones. Our trial results are shown in figure 3; a series of trial were done, four selected results are shown, in different conditions; the extracted data after iterative method is displayed. The trials now are in their first step and will be continued for two or three months, after that results will be discussed to develop the prototype and add the appropriate modification on processing program.
Figure 3 four trial results

Figure 4 shows a comparisons between two processed signals for the same data frame; the first with classical detection methods, and the second with iterative detection method. It is clear in this figure the presence of false target in classic detection method.

Conclusion Underwater intruders are acoustically quiet; their detection often relies on target echoes using broadband. A monitoring system approach discussed in this paper, considering the site investigation, using combining equipment types such as sonar, hydrophones, communication tools, multithreading software programs and so on. As we mentioned the simulation results for the zero version were satisfied. The results shown in figure 3 are showing promising mechanism for detecting the underwater intruders. We believe that the improved next version with precise analysis and more threshold study will give more idea about the effectiveness of this mechanism. The prototype trial result gives us promotion to continue in our approach to have integrated anti intruder monitoring system. We have detected some misleading result in our method. We are now making a series of trials, and according to these results we will assess the iterative method efficiency. After that we are intending to modify our system to have best results for intruders’ detection system. We hope the situation in Syria will be better to make more trials and researches.
REFERENCES

THE MAIN PECULIARITIES OF AUTOMATIC TARGET RECOGNITION IN UNDERWATER ACOUSTICS

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Abstract: The special features of the underwater automatic target recognition (ATR) problem are examined. The general approach to the synthesis of the optimal maximum likelihood underwater ATR algorithm is discussed. The approach is illustrated by the example of the submarines and surface ships recognition at the output of passive sonar.

Keywords: underwater acoustics, automatic target recognition, optimal recognition algorithms.
1. INTRODUCTION

Automatic target recognition (ATR) is one of the most challenging problems of ap-
plied underwater acoustics.

Underwater ATR as science appeared at the beginning of the 60's of past century. For
a long time the target recognition was considered to be the skill of rumor operator to rec-
ognize noise and echoes. The need for the development of objective methods of the target
recognition arose in the end of the 50's of past century in connection with the appearance
of new sonar in which the signal integration was realized. That has caused sharp decrease
of the signal-to-noise ratio (SNR) threshold and growth of the target detection range. As a
result the sonar operator began to experience difficulties with target recognition at large
detection distances.

Underwater targets are recognized using the target signal parameters (TSP), which
possesses information about the target class and can be measured at the sonar receiver
output.

The purpose of the paper is to discuss the main peculiarities of the ATR problem and
to propose a general approach to the synthesis of optimal ATR algorithms which can be
applied to both passive and active sonar.

1. THE MAIN PECULIARITIES OF THE ATR PROBLEM

The main peculiarities of the ATR problem are the following:
- ATR is based on 3 types of TSP caused by the peculiarities of target class:
  - sound generation (or sound reflection);
  - underwater sound propagation;
  - behavior;
- since the ATR is fulfilled as a rule at small signal-to-noise ratios (SNR) the TSP
do not possess much useful information and that is why the only way to reach high effec-
tiveness of ATR is correct usage of all measured TSP;
- the information concentrated practically in each TSP depends on the current hydro
  acoustic conditions and SNR, that is why ATR algorithms must be adaptive;
- practically all TSP are mutually dependent/ It dictates the need to base the ATR
  algorithms on mutual probability distribution function (PDF) of TSP;
- different TSP are measured during different time intervals, that is why ATR algo-
  rithms must be dynamic.

The experience shows that the lack of knowledge about the above-mentioned peculi-
arities does not allow the ATR problem to be solved with the required quality. For exam-
ple, the algorithms which do not consider current hydro acoustic conditions and also those
in which the recognition decisions are delivered with the use of separate TSP with subse-
quent weight summing of these decisions are ineffective.

2. THE PROCEDURE OF THE OPTIMAL ATR ALGORITHM SYNTHESIS

Let us state the formulation of the problem. Assume that they are assigned:
- the alphabet (the vector) $\Omega$ containing $m$ target classes which form the full group
  of the events (i.e. each target which may be detected corresponds in the alphabet it's own
class but only one):
\[ \Omega = \{ \omega_1, \omega_2, \ldots, \omega_m \} \]  
- the vector \( \mathbf{X} \) containing \( n \) TSP \( X_s \): 
\[ \mathbf{X} = \{ X_1, X_2, \ldots, X_n \} \]

It is necessary to synthesize the algorithm, which ensures the best ATR from the point of view of the selected statistical criterion.

It is known [3] that the full class of algorithms for the formulated problem (the class which contains all optimal algorithms of multi-class recognition) has the form

\[ \omega_{\text{opt}} = \arg \min_j \left\{ R(\omega_j, \hat{X}) \right\} \]  
(3)

where \( \omega_{\text{opt}} \) is a result of the recognition (the optimal target class); \( \hat{X} \) is the estimate of TSP vector \( \mathbf{X} \); \( R(\omega_j, \hat{X}) \) is the Bayesian risk function (BRF) which equals the loss of the decision in favor of the \( j \)th class (with the use of the vector \( \hat{X} \)):

\[
R(\omega_j, \hat{X}) = \frac{\sum_{i=1}^m P(\omega_i) \cdot C_{i,j} \cdot g_{X/\omega_i}(\hat{X})}{\sum_{i=1}^m P(\omega_i) \cdot g_{X/\omega_i}(\hat{X})};
\]  
(4)

\( P(\omega_i) \) is the apriori probability of \( i \)th class target detection; \( C_{i,j} \) is the cost of the false decision when the target of the \( i \)th class was classified as \( j \)th class; \( g_{X/\omega_i}(\mathbf{x}) \) is conditional probability density distribution (PDD) of the TSP vector \( \mathbf{X} \) estimate which converts into the likelihood function (LF) of the target class when the nonrandom argument \( \mathbf{x} \) is changed by random TSP vector \( \mathbf{X} \) estimate \( \hat{X} \) [4].

According to the formula (3) the optimal class \( \omega_{\text{opt}} \) minimizes the value of the LF which is calculated as the linear combination of \( m - 1 \) alternative classes. The coefficients in the linear combination are multiplications of the apriori target detection probability and the cost of the false decision.

For the concrete definition of the class of algorithms (3) in our case let us take into consideration the fact that the parameters \( P(\omega_i) \) and \( C_{i,j} \) are unknown. Many attempts to motivate them failed without result due to the complexity of this task, caused by the need to take into account the large number of random factors. But it mustn't disturb the synthesis of the optimal recognition algorithms because according to a well known postulate [3] in a real information systems the apriori determined parameters must not exert a substantial influence on the solutions. Taking into account that in practice the smaller the apriori target detection probability the larger the cost of its false recognition (i.e. the less frequently the target is detected, the more it is available), it is possible to allow the validity of the condition

\[
P(\omega_i) \cdot C_{i,j} \approx \begin{cases} \text{const} & \text{if } i \neq j \\ 0 & \text{if } i = j \end{cases}
\]  
(5)

The substitution of (5) in (4) leads algorithm (3) to the form

\[ \omega_{\text{opt}} = \arg \max_j \left\{ P_{\text{apost}}(\omega_j, \hat{X}) \right\} \]  
(6)

where \( P_{\text{apost}}(\omega_j, \hat{X}) \) is the aposteriori probability of belonging the target to \( j \)th class, which can be evaluated as follows [4]
Thus, the algorithm (6) is the optimal ATR algorithm. The probability weight of the optimal decision can be evaluated with help of formula (7) with the substitution $\omega_j = \omega_{opt}$.

Let us note that formula (7) is Bayes' formula for the case of the equal *apriori* target detection probabilities of each class, and algorithm (6) realizes the maximum likelihood method (ML-method) widely utilized in image recognition theory.

If one needs to introduce into the recognition algorithm the zone of failure the algorithm (6) is converted to the form:

$$K = \begin{cases} \omega_{opt}, & \text{if } P_{apost} (\omega_{opt}, \hat{X}) > P_{thresh} \\ \text{refuse}, & \text{if } P_{apost} (\omega_{opt}, \hat{X}) \leq P_{thresh} \end{cases}$$

(8)

where $P_{thresh}$ is the threshold probability.

Let us view the organization of the target recognition. From the moment of target detection the cyclical measuring process of target TSP starts. The time of measurement of each TSP is individual. Simultaneously with the TSP measuring process the cyclical process of making decisions according the algorithm (8) starts. The cycle time of this process, as a rule, coincides with the time of the very “rapid” cycle of TSP measurement. Since TSP measurement and making of a decision realize asynchronously, the TSP amount utilized at the different cycles of decision can differ.

To increase the statistical regularity of the decisions the measurements of the similar TSP, obtained on the sequential cycles, are smoothed by one of the methods. The *aposteriori* probabilities, used in the algorithm (6), can also be smoothed. Usually this is achieved by the $\alpha$-filter application:

$$P_{apost/smooth} (\omega_j) = \alpha \cdot P_{apost} (\omega_j, \hat{X}) + (1 - \alpha) \cdot P_{apost/smooth} (\omega_j), \quad j = 1, ..., m$$

where $P_{apost} (\omega_j, \hat{X})$ is the *aposteriori* probability of belonging the target to $j$th class, evaluated with help of formula (7) at current decision cycle; $P_{apost/smooth} (\omega_j)$ is the smoothed *aposteriori* probability of belonging the target to $j$th class; $\alpha$ is the constant which controls the smoothing time ($0 < \alpha \leq 1$).

### 3. THE CALCULATION OF THE MUTUAL CONDITIONAL PDD OF TSP ESTIMATES

From the revue of formulas (5)-(8) it follows that the only nontrivial operation in the procedure of ATR algorithm synthesis is the calculation of the conditional (depending on the target class) PDD $g_{X_i/X_j}$ of the TSP vector estimate $X$. Therefore let us pay more attention to this question.

For the PDD $g_{X_i/X_j} (x)$ calculation the following procedure was developed.

1) For each TSP $X_i$, belonging to the vector $X$, the stochastic model is created in the form:
\[ \hat{X}_s = \varphi_s(Z_s) + \Delta X_s \]  

(9)

where \( \hat{X}_s \) is an estimate of TSP \( X_s \) (scalar or vector); \( \varphi_s(Z_s) \) is the nonrandom function connecting the TSP \( X_s \) true value with the target sound emission (or reflection) parameters, the target coordinates and moving parameters and the signal channel propagation parameters, i.e.,

\[ X_s = \varphi_s(Z_s). \]  

(10)

Functions \( \varphi_s(Z_s) \) make it possible to incorporate the effects of hydroacoustic conditions into the recognition algorithm; \( \Delta X_s \) is an estimation error of TSP \( X_s \). The error statistical characteristics (for example, the variance) allow inclusion of noise conditions into recognition algorithms.

The \( Z_s \) vectors for different TSP can contain the different quantity of parameters.

2) The vector \( Z \), which includes all parameters belonging at least to one vector, is formed.

3) The mutual conditional (depending on the target class and \( Z \) vector) PDD of all TSP \( \hat{X}_s \) is calculated. Since the TSP estimation errors \( \Delta X_s \), as a rule, are independent, all TSP estimates \( \hat{X}_s \) become independent too. Therefore their mutual conditional PDD equals to the multiplication of conditional PDD of every TSP [5], i.e.

\[ g_{\hat{X}_s/Z} (x) = \prod_{s=1}^{n} g_{\hat{X}_s/Z_s} (x_s) = \prod_{s=1}^{n} g_{\Delta X_s} (x_s - \varphi_s(Z_s)) \]  

(11)

4) The mutual conditional (depending only on the target class) PDD of all TSP estimates \( \hat{X}_s \), united in the vector \( \hat{X} \), is calculated [3]:

\[ g_{\hat{X}/a} (x) = \int_{z} \int_{z} g_{\hat{Z}/a} (z) \cdot g_{\hat{X}/z} (x) \cdot dz \]  

(12)

where \( g_{\hat{Z}/a} (z) \) is mutual conditional PDD of the parameters, forming the vector \( Z \).

Since the majority of these parameters are mutually independent, the PDD \( g_{\hat{Z}/a} (z) \) is calculated as the multiplication of the PDD of each vector \( Z \) element.

4. EXAMPLE OF THE OPTIMAL ATR ALGORITHM SYNTHESIS

Let us examine a simple but practically important example for the passive sonar. Let the alphabet of recognized classes \( \Omega \) includes only 2 classes: submarine (SM) and surface ship (SS). And let the TSP vector \( X \) contains 2 elements:

- \( SL \) – the target signal level in a certain wide frequency band;
- \( BD \) – the target bearing derivative (or rate).

These TSP are widely used by sonar operators: when the detected target has small \( SL \) and large \( BD \) it is named SM; and vice versa, when the detected target has large \( SL \) and small \( BD \) it is named SS.

The stochastic model of SL estimate has the form:

\[ \hat{SL} = SL(V, R, \Delta D) + \Delta SL \]  

(13)

where \( \hat{SL} \) is SL estimate; \( SL(V, R, \Delta D) \) is nonrandom function which connects the SL true value with the true values of target speed \( V \), target range \( R \) and the calculation error
This function relates the current conditions to the recognition algorithm [6]; $\Delta SL$ is the SL estimation error.

The stochastic model of BD estimate has the form:

$$\hat{BD} = BD(V, C, R) + \Delta BD$$  \hspace{1cm} (14)

where $\hat{BD}$ is BD estimate; $BD(V, C, R)$ is nonrandom function connected the true BD value with true values of target speed, course and range; $\Delta BD$ is a BD estimation error.

According to the formulas (13) and (14) the vector $Z$ must include 4 unknown parameters – target speed, course, range and SL calculation error:

$$Z = \{V, C, R, \Delta D\}$$  \hspace{1cm} (15)

Taking into account the independence of the TSP estimation errors, the mutual conditional (if the vector $Z$ value is fixed) PDD of SL and BD estimates can be calculated as follows:

$$g_{SLBD\mathbb{Z}}(s, b / z) = g_{SL\mathbb{Z}}(s / z) \cdot g_{BD\mathbb{Z}}(b / z) =$$

$$g_{SL\mathbb{Z}}(s - SL(V, R, \Delta D)) \cdot g_{BD}(b - (V, C, R))$$  \hspace{1cm} (16)

where $g_{SL\mathbb{Z}}(s / z), g_{BD\mathbb{Z}}(b / z)$ are conditional (as a function of vector $Z$) PDD of SL and BD estimates; $g_{SL\mathbb{Z}}(s), g_{BD}(b)$ are SL and BD estimation errors PDD. These PDD as a rule have Gaussian form with zero mean and variance depending on SNR:

$$g_{SL\mathbb{Z}}(s) = \text{norm}(s / 0, \sigma_{SL}^2)$$

$$g_{BD\mathbb{Z}}(b) = \text{norm}(b / 0, \sigma_{BD}^2)$$

$\text{norm}(x / \mu, \sigma^2)$ is Gaussian PDD of with argument $x$, mean value $\mu$ and variance $\sigma^2$; $\sigma_{SL}^2$ and $\sigma_{BD}^2$ are variance of the $\Delta SL$ and $\Delta BD$ estimation errors.

To transform the mutual PDD, depending on vector $Z$, to the mutual PDD, depending on target class, one must integrate the PDD (16):

$$g_{SLBD\omega_{at}}(s, b / v, c, r, d) = \int \int \int g_{Z\omega_{at}}(v, c, r, d) \cdot g_{SLBD\mathbb{Z}}(s, b / v, c, r, d) \cdot dv \cdot dc \cdot dr \cdot dd$$  \hspace{1cm} (17)

where $g_{Z\omega_{at}}(v, c, r, d)$ is mutual conditional PDD of vector $Z$.

Since the parameters, belonging to vector $Z$, are independent, the mutual conditional PDD of vector $Z$ can be written as

$$g_{Z\omega_{at}}(v, c, r, d) = g_{v\omega_{at}}(v) \cdot g_{c\omega_{at}}(c) \cdot g_{r\omega_{at}}(r) \cdot g_{d\omega_{at}}(d)$$  \hspace{1cm} (18)

Fig.1 shows the ATR algorithm block diagram, corresponding to one decision cycle.

The effectiveness of the discussed algorithm is shown on the Fig.2. The axes represent the estimates of the target signal level ($SL$) in a certain wide frequency band and the target bearing rate ($BD$). The color corresponds (see the bar right wise the figure) to the $\text{aposteriori}$ probability of the target class decisions $P_{\text{apost}}(\omega_{\text{opt}}, \hat{SL}, \hat{BD})$, delivered according the formula (8). The threshold probability $P_{\text{thresh}}$ in this example equals 0.6. Fig.2a corresponds to the distance of target detection (SNR=-15 dB at linear output of the antenna), Fig.2b – to the distance of target tracking (SNR=-5 dB).

The results of the Fig.2 analyses are the follows:

- at the target detection distance we see large space of parameter values (in the left lower corner) where the recognition effectiveness is small. The effective recognition can
be fulfilled if the SL value is greater than –3 dB or if the BD value is greater than 0.16 degree per minute;
- at the target tracking distance (which equals 0.5–0.8 of the detection distance) the non effective space is more narrow but does not disappear at all;
- to increase the recognition effectiveness one should use some extra signal parameters.

Fig.1. The ATR algorithm block diagram, corresponding to one decision cycle
5. SUMMARY

The paper contains the brief revue of general approach to the underwater automatic target recognition (ATR) algorithm syntheses. This approach is based on the main features of ATR problem and can be used to design the modern passive and active sonar.

6. ACKNOWLEDGEMENTS

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INDEPENDENT VIEWS IN MIMO SONAR SYSTEMS

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\textbf{Abstract}: The main advantages of MIMO (Multiple Input Multiple Output) sonar systems come from the assumption of independent observations between each transmitter/receiver pairs. The independence of the observations ensures a unbiased set of measurements and then provides true statistics on the target. In this paper we study the correlation between views in MIMO sonar systems. A traditional tool used to study the dependency between two random variables is the Pearson product-moment correlation coefficient. However this measure suffers numerous defaults: it only estimates linear correlation, it is not a proper distance and in particular a null measure of the Pearson coefficient does not insure the independence of the tested random variables. For these reasons we will use the distance correlation introduced by Szekely. From the distance correlation we will derive the inter-views distance correlation matrix which assess the correlation of the full MIMO system (i.e. the dependencies between each views). This independence measure matrix gives a guideline to how to build truly uncorrelated MIMO sonar systems and then maximise the performances of such system.

\textbf{Keywords}: MIMO sonar systems, Multi-static sonars.
1. INTRODUCTION

MIMO stands for Multiple Inputs Multiple Outputs. It refers to a structure with spatially spaced transmitters and receivers and has huge implications in wireless communications mainly to overcome the multipath problem in complex urban environments. MIMO sonar systems had raised a lot of interests for ASW purposes. In previous papers [1, 2] we formalised the MIMO theory for wide and narrow band sonar systems. Assuming independent observations between each transmitter/receiver pairs we demonstrated two important results for MIMO systems:

1. it is possible to evaluate the scatterer density of a target if the number of scatterers is low.

2. with a sufficient number of views MIMO systems can decorrelate the scatterers contributions within one pixel resolution and then resolves the speckle.

Result 1. is archived by computing the probability \( P(T|Q) \) as a function of \( P(\Gamma|T_Q) \) using Bayes rules where \( \Gamma = \{\gamma_n\}_{n \in [1,N]} \) is the set of observations and \( T_Q \) the event of observing a target \( T \) with \( Q \) scatterers. Assuming the independence of the observations allows us to write:

\[
P(\Gamma|T_Q) = \prod_{n=1}^{N} P(\gamma_n|T_Q)
\] (1)

Intuitively we sense that as the total number of views increases \( \Gamma \) tends toward the PDF of the target intensity echo. In the extreme case where all the views are correlated we have \( P(\Gamma|T_Q) \approx P(\gamma_n|T_Q) \) so no ID information can be extracted from \( \Gamma \).

Result 2. comes from the asymptotic behaviour of the PDF of the detection rule \( F(r) \sim \frac{1}{N} \sum_{n=1}^{N} \text{Rayleigh}^2(\sigma) \). The independence of the observations allows us to write

\[
\frac{1}{N} \sum_{n=1}^{N} \text{Rayleigh}^2(\sigma) \sim N.\Gamma(Nx,N,1)
\] (2)

and then to deduce that:

\[
\lim_{N \to +\infty} N.\Gamma(Nx,N,1) = \delta(1-x)
\] (3)

Again the independence assumption allows the factorisation of the sum of the squared Rayleigh distributions. If we consider the extreme case where all the observations are dependent we would have: \( \frac{1}{N} \sum_{n=1}^{N} \text{Rayleigh}^2(\sigma) \sim \text{Rayleigh}^2(\sigma) \) and then the speckle effect still dominates the pixel intensity behaviour.

In this paper we study the correlation between views in MIMO sonar systems using the distance correlation introduced by Székely [3]. From the distance correlation we will derive the inter-views distance correlation matrix which assess the correlation of the full MIMO system (i.e. the dependencies between each views). This independence measure matrix gives a guideline to how to build truly uncorrelated MIMO sonar systems and then maximise the performances of such system.
2. THE INDEPENDENT VIEWS PROBLEM

2.1. Definition

We define independent views as: two views are independent if and only if their observations are statistically independent.

2.2. The distance correlation

By introducing the term view we implicitly introduce the geometry and the configuration of the MIMO system. Let θ be the target view angle of the transmitter and φ the view angle of the receiver. The bistatic configuration of a pair of transmitter/receiver will be noted (θ, φ). We are interested here in knowing the level of independence of a view \( V(\theta_1, \phi_1) \) with another view \( V(\theta_2, \phi_2) \). To measure the dependence of 2 random variables the Pearson product-moment correlation coefficient or correlation coefficient is commonly used [4]. However the correlation coefficient suffers a number of defects: First of all this coefficient has been designed with a normal distribution assumption, this assumption does not hold in our case. Secondly this coefficient only measures linear correlation between the random variable. And finally this coefficient is not a real independence measure in the sense that the correlation coefficient of 2 random variables can be null even if these random variables are dependent. To overcome these defects we will be using the distance correlation introduced by Székely in [3]. Székely defines the distance covariance \( \mathcal{V} \) as:

\[
\mathcal{V}^2 = \frac{1}{c_p c_q} \int_{\mathbb{R}^{p+q}} \left| f_{X,Y}(t,s) - f_X(t) f_Y(s) \right|^2 \frac{dt ds}{|t|^{1+p}|s|^{1+q}}
\]

(4)

where \( f_X \) and \( f_{X,Y} \) represent respectively the characteristic and the joined characteristic function of \( X \) or \( (X,Y) \), \( p \) and \( q \) are respectively the dimensions of the random vector \( X \) and \( Y \), and \( c_d \) is defined as follow:

\[
c_d = \frac{\pi^{(1+d)/2}}{\Gamma((1+d)/2)}
\]

(5)

where \( \Gamma(.) \) is the full gamma function. For \( \mathcal{V}^2(X) \mathcal{V}^2(Y) \neq 0 \) the distance correlation is then defined as:

\[
\mathcal{R}^2(X,Y) = \frac{\mathcal{V}^2(X,Y)}{\sqrt{\mathcal{V}^2(X)\mathcal{V}^2(Y)}}
\]

(6)

Székely shows in [3] that \( \mathcal{R} \) has the properties of a true dependence measure and in particular that two random vectors \( X \) and \( Y \) are independent if and only if \( \mathcal{R}(X,Y) = 0 \).

2.3. MIMO inter-views dependence

To assess the inter-views dependence of a MIMO system we generate randomly \( 10^4 \) targets with 2, 3, 4 or 5 scatterers. For each target we compute its response \( V \) as a function of the transmitter and receiver view angle \( (\theta, \phi) \). For each pair \( (\theta, \phi) \), \( V(\theta, \phi) \) can then be considered
as a random vector. The distance correlation $R$ between all pairs $(\theta_n, \phi_n) \in [-\pi, \pi]^2$ is then computed. For the view angles $(\theta_0, \phi_0)$, let $A_0$ be the matrix defined by:

$$A_0(\theta, \phi) = R(V(\theta_0, \phi_0), V(\theta, \phi))$$

(7)

Note that in the point scatterer model there is a symmetry between the transmitter and the receiver and $V(\theta, \phi) = V(\phi, \theta)$. For this reason the matrix $A_0$ is symmetric along its first diagonal.

Let $\theta_1 = \theta_0 - \alpha$ and $\phi_1 = \phi_0 - \alpha$. Thanks to the axial symmetry of the problem we can write that:

$$A_0(\phi, \theta) = A_1(\phi - \alpha, \theta - \alpha)$$

(8)

So we can compute $A_0(\theta, \phi)$ for only one $\theta_0$. We then can chose $\theta_0 = 0$. For display purposes we display in Fig. 1 the distance correlation matrix $1 - A_0(\theta, \phi)$ for $\phi_0 = 0$, $\phi_0 = \pi/2$ and $\phi_0 = \pi$.

![Fig. 1: Distance correlation matrix $1 - A_0(\theta, \phi)$ for (a): $\phi_0 = 0$, (b): $\phi_0 = \pi/2$ and (c): $\phi_0 = \pi$.](image)

Fig. 1 displays the monostatic case, the transmitter and the receiver are in the same position: $\theta_0 = \phi_0 = 0$. Even though the monostatic configuration is convenient from a practical point of view it does not offer the best view in term of correlation. The monostatic view correlates strongly with its neighbours $(\theta = +\alpha, \phi = -\alpha)$ for $\alpha \in [-25^\circ, +25^\circ]$. It is interesting to note that the monostatic view correlates as well with $(\theta = \alpha, \phi = \alpha)$ for $\alpha \in [-6^\circ, +6^\circ]$. So if we consider a monostatic sonar turning around the target for a full $360^\circ$, an average around 30 independent views will be obtained which is insufficient to achieve super-resolution. In Fig. 1(c) the target is in-between the transmitter and the receiver. Although this configuration is not practical as the transmitted wave will arrive at the same time as the target echo to the receiver, it is interesting to note that all the opposite views $(\theta, \theta + \pi)$ for all $\theta$ correlate strongly. In Fig. 1(b) displays the distance correlation matrix with $\phi_0 = \pi/2$. As predicted we observe a symmetry along the first diagonal and $A_0(\theta, \phi) = A_0(\phi, \theta)$. The correlation peaks are focused on $(\theta_0, \phi_0)$ and $(\phi_0, \theta_0)$. This configuration is the most effective as far as its independence is concerned. And the independence of this view toward its neighbours is maximised.
2.4. MIMO geometry and super-resolution

In the following simulation we aim to demonstrate that we can recover the geometry of a target (i.e. the location of its scatterers). Given the results presented in Fig. 1 we chose a "L" shape MIMO configuration as pictured in Fig. 2. The transmitters are placed in the x-axis, the receivers are on the y-axis. For this experiment we place the transducers an equal spacing along the axis. The number of transmitters and receivers and the spacing between them is adjustable. The MIMO system will use the frequency band 50 kHz to 150 kHz. We consider a 3 point scatterers target, the scatterers are separated by one wavelength. Note that we are considering the central frequency of the MIMO system (100 kHz). Under this condition one wavelength corresponds to 1.5 cm.

In order to image the output of the MIMO system we will use the multi-static back-projection algorithm which is a variant of the bistatic back-projection algorithm developed by the Synthetic Aperture Radar (SAR) community. Further details can be found in [5–7]. Using the back-projection algorithm the Synthetic Aperture Sonar (SAS) image is computed by integrating the echo signal along a parabola. In the bistatic case the integration is done along ellipses. For the multi-static scenario the continuous integration is replaced by a finite sum in which each term corresponds to one transmitter/receiver pair contribution.

![Fig. 2: MIMO configuration.]

In Fig. 3(a) we have considered a MIMO system with 10 transmitters and 10 receivers with a spacing of 20 cm. For this scenario the 20 cm spacing breaks the widely spaced antenna
assumption and the views are not exactly independent between each other. For this reason we only observe a blob of energy at the target location. In Fig. 3(b) the MIMO system consists in 3 transmitters and 3 receivers with 3 metres spacing. In this case the spacing between the antennas is several hundreds of wavelengths so the independence of the views is respected. The total number of views however is $3 \times 3 = 9$ independent views which is relatively low according to the convergence speed of Eq. (3). In this scenario the number of views is too low to ensure the decorrelation of the scatterers within the target. For this reason only a blob of energy marks the target location. However by closely inspecting to the central blob it is possible to distinguish a structure. Finally in Fig. 3(c) we consider a MIMO system with 10 transmitters and 10 receivers with a spacing of 3 metres. With this configuration we respect the conditions stipulated earlier and we are able to clearly image the 3 scatterer target in so doing achieve super resolution imaging.

### 2.5. Inter-correlation for full MIMO systems

It is interesting to compare these results to the intra-views correlation of the different MIMO systems. Let note $\{(\theta_n, \phi_n)_{n \in [1,N]}\}$ the views of the MIMO system. The level of inter-correlation for the full MIMO can be computed as:

$$B(\theta, \phi) = \max_{n \in [1,N]} A_n(\theta, \phi)$$

\[(9)\]

In Fig. 4 we plot the $1 - B(\theta, \phi)$ functions for the same MIMO configurations as the ones explained in Fig. 3. In Fig. 4(a) we are considering the $10 \times 10$ MIMO system with 20 cm separation between antennas. The 100 views produced by this configuration are all concentrated around the $(0^\circ, -90^\circ)$ view and are clearly all correlated to each other. The independent views assumption breaks down. In Fig. 4(b) the $3 \times 3$ MIMO configuration is considered. The 3 m spacing between the antenna ensures the independence of the views and we can clearly see in the cluster 9 peaks corresponding to each view. In Fig. 4(c) the $10 \times 10$ MIMO configuration is considered. Again the 3 m antenna separation provide the necessary independence between the views and the 100 correlation peaks are visible and distinct between each other. The $B(\theta, \phi)$ inter-correlation distance matrix then gives us an insight on how to design an efficient MIMO.
system and ensure the views independence. Assuming that the MIMO system provides enough views for recognition or super-resolution, each views \( (\theta_n, \phi_n) \) in the \( B(\theta, \phi) \) should decorrelate as much as possible with the other views \( (\theta_m, \phi_m)_{m \neq n} \)

For comparison purposes we have computed the SAS image of the same target as described in Fig. 2 using the same frequency band and at the same range than the previous experiment. The SAS image of the target is displayed in Fig. 5.

![SAS Image](image)

**Fig. 5:** 3 scatterers target using SAS system. (a) SAS image, (b) \( 1 - B(\theta, \phi) \) function for the SAS configuration.

The SAS system run at 20m range from the target in a straight line. The beamwidth is fixed to 10°. We compute the target echo at every \( \lambda/2 \). In total 467 echoes are computed and the SAS image is formed using back-propagation algorithm. Despite the high number of views and because all the SAS subviews are highly correlated between each other as shown in Fig. 5(b), the SAS system fails to separate the 3 scatterers. Earlier we saw that monostatic systems correlate in average for 12°. A 10° beamwidth SAS system then sees at the maximum 2 to 3 independent views of a target. Note that on this aspect the SAS image reconstruction is based on the hypothesis that each pixel contains one scatterer point. And then SAS systems requires strong correlation between consecutive views in order to track and correct the echoes phase changes. So in that aspect it is not surprising that the mono-views from SAS systems are so strongly correlated to each other.

3. CONCLUSION

In this paper we were able to relate independent observations to independent views for MIMO systems. We derived the MIMO inter-views dependence and then the full MIMO system inter-correlation factor using the distance correlation introduced by Székely. We showed that the 90° bistatic view is the one which correlates the less to the other views and the one which is the more compact in the MIMO inter-correlation distance plane. With those results a fully independent inter-views MIMO system was designed and we demonstrated that such system has the predicted MIMO capability especially such system can archive super resolution.
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AUTOMATIC CLASSIFICATION FOR MID-FREQUENCY ANTI-SUBMARINE WARFARE SONARS - RECOGNIZING PIPELINES

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Abstract: The Norwegian Trench is practically littered with wrecks, pipelines, and underwater terrain features, all capable of generating false contacts in anti-submarine warfare (ASW) sonars. For modern sonar systems, the sonar operator may easily discard such false alarms simply by overlaying the detections on a chart containing the positions of pipelines and wrecks. However, autonomous systems lack human perception and intelligence, and therefore require failsafe algorithms to filter out false contacts.

Here we study the theoretical behavior of echoes from pipelines and also derive an algorithm for classifying a given detection as originating from a pipeline or not. The method is applied on track level data and exploits the kinematical properties of tracks to estimate a probability that a given track originates from a given pipeline. A modification to the algorithm that allows for accurate classification in unchartered waters is also suggested.

The proposed method is tested on a data set called the Clutter Experiment 2002 (CEX02). This experiment was carried out in the NAT3 programme, a collaboration between Thales Underwater Systems, TNO, FFI, and the Dutch, French, and Norwegian navies, with the intent to experiment with and evaluate different clutter-reducing techniques. The area of the Norwegian Trench where the test was carried out is particularly difficult with regards to false alarms, due to strong upslopes and the presence of both pipelines and seamounts.

Keywords: Sonar, classification, pipelines
INTRODUCTION

Operation of anti-submarine warfare (ASW) sonars in littoral waters typically results in high false alarm rates due to both clutter and the presence of strong reflectors such as large rocks, ship wrecks, and pipelines. Classification based on features extracted from received echoes is a possible solution to this problem. For conventional ASW sonar systems the sonar operator typically classifies a given echo as either false or a possible underwater threat, for instance by listening to the echo or visually studying the echogram. However, automated sonar systems, e.g. automated underwater vehicles (AUV) equipped with sonars, do not have the luxury of human intelligence and perception. Such systems require failsafe, automatic classification algorithms.

Most sources of false alarms in the underwater battle space are stationary. The kinematical properties of tracks [1] are therefore excellent classification features. Unfortunately, due to the highly directional target strength of pipelines, they appear as moving targets to a sonar. This makes the kinematical properties insufficient to determine if a given track is on an enemy submarine or a pipeline. However, pipelines make highly predictable targets. Here we propose a method that exploits this predictability to distinguish between tracks on pipelines and real targets. The method is tested on a data set collected in the NAT III (New array technology III) cooperation in 2002. The partners of this cooperation included TNO, Thales Underwater Systems, FFI, and the French, Dutch, and Norwegian navies.

THEORY

Pipelines are long cylindrical objects with extremely high and directional target strength. Due to the high directivity, the vector that connects the sonar to a pipeline echo is normal to the main axis of the pipeline, as illustrated in Fig. 1.

Given the position of the pipeline and sonar, estimating the expected position of the generated echo in the /th transmission, $\hat{x}_{ej}$, is straightforward:

$$\hat{x}_{ej} = x_p + (r_j \cdot p)p$$

$x_p$ is an arbitrary position along the pipeline. $r_j$ is a vector connecting the sonar position, $x_{sj}$ to $x_p \cdot p$ is a unit vector describing the direction of the main axis of the pipeline. The measured echo position, $x_{ej}$, has an associated measurement error:

$$\Delta r_j = r_j - \hat{r}_j$$

$$\Delta \alpha_j = \alpha_j - \hat{\alpha}_j$$

The polar coordinates used are explained in Fig. 1. The estimated and measured velocity may be determined from two successive transmissions and have errors given by:

$$\Delta u = u_j - \hat{u}_j$$

$$\Delta \theta_j = \theta_j - \hat{\theta}_j$$

where $u_j$ and $\theta_j$ are the velocity magnitude and bearing, respectively.
Consider a track with $J$ track updates and positions given by $x_{ej}$. The problem at hand is to determine whether the given track is caused by a present pipeline (hypothesis $H_1$) or not (hypothesis $H_0$). The Neyman-Pearson lemma states that:

$$L(x_{ej}) = \ln \frac{P(x_{ej}|H_1)}{P(x_{ej}|H_0)}$$  \hspace{1cm} (6)$$

The nominator is the probability that the measurement, $x_{ej}$, is made when $H_1$ is true, while the denominator is the probability that $H_0$ is true. Assuming Gaussian distributed and independent errors, $P(x_{ej}|H_1)$ may be expressed by:

$$P(x_{ej}|H_1) = \frac{1}{(2\pi)^2\sigma_r\sigma_\alpha\sigma_u\sigma_\theta} \exp \left( -\frac{1}{2} \left( \frac{\Delta r_j^2}{\sigma_r^2} + \frac{\Delta \alpha_j^2}{\sigma_\alpha^2} + \frac{\Delta u_j^2}{\sigma_u^2} + \frac{\Delta \theta_j^2}{\sigma_\theta^2} \right) \right)$$  \hspace{1cm} (7)$$

where $\sigma_r$, $\sigma_\alpha$, $\sigma_u$, and $\sigma_\theta$ are standard deviations of the random errors. Note that the assumption of independent errors does not really apply, since measured velocity is derived directly from the positions. However, some simplifications are needed to derive a manageable model for this problem. Assuming all bearings and ranges have equal probability of generating a non-pipeline detection, then $P(x_{ej}|H_0)$ is constant.

Whether the hypothesis $H_1$ or $H_0$ is true is typically decided by comparing Eq. (6) to a threshold. The choice of this threshold is important, because by setting the threshold too low, then non-pipeline tracks may be interpreted as pipeline tracks (false alarms), while a too high threshold would result in the opposite (missed classifications). Applying a threshold yields the following for the case where $H_1$ is true:

$$\frac{\Delta r_j^2}{\sigma_r^2} + \frac{\Delta \alpha_j^2}{\sigma_\alpha^2} + \frac{\Delta u_j^2}{\sigma_u^2} + \frac{\Delta \theta_j^2}{\sigma_\theta^2} \leq T$$  \hspace{1cm} (8)$$

otherwise $H_0$ is considered true. $T$ is the selected threshold. The standard deviations are not easily determined. E. g., the range error depends on the error of other parameters such as arrival time, sound speed, own ship position, time synchronisation between transmitter.
and receiver, etc. We suggest estimating the standard deviations using a training data set. The training data set contains both pipeline and non-pipeline tracks that are first classified by hand. The standard deviations and threshold are then adjusted in order to perfectly classify all the tracks. A test data should then be used to validate the chosen parameters.

The positions of pipelines are not always known. The proposed method may be extended to apply also for an unknown position. In that case, the position of the pipeline must be estimated from the track itself. This is easily done by applying a sliding averaging window of width $2N+1$ over the track positions:

$$\bar{x}_{pj} = \frac{1}{\sum_{k=-N}^{N} G(k)} \sum_{k=-N}^{N} G(k) \bar{x}_{ej}$$

(9)

where $G(k)$ is a weighting function, e.g. uniformly or Gaussian shaped weights. The estimation of the pipeline position comes at the cost of two of the four classification features used in Eq. (8); $\Delta r_j$ and $\Delta \alpha_j$. These two errors are low since the pipeline positions are estimated from the track positions. However, the speed and course errors are still valid and useful for classification purposes, reducing Eq. (8) to:

$$\frac{\Delta u_j^2}{\sigma_u^2} + \frac{\Delta \theta_j^2}{\sigma_\theta^2} \leq T_{NP}$$

(10)

where the new threshold $T_{NP}$ should be selected in the same manner as $T$.

An issue with this approach is that in order to get a decent estimate of the pipeline position from the track positions, the track has to be of sufficient length. Here we require a track to be at least ten transmissions long in order to be classified. This causes a latency which may be problem in an operational setting. The approach where the pipeline positions are known do not have an issue with track lengths and latency.

**EXPERIMENTAL DATA**

The proposed method was tested on a data set collected during CEX02 (Clutter Experiment 02) the NAT III (New Array Technology III) trial in September 2002. This data set has also been used in earlier open publications [2]. The experimental setup consisted of two receiver arrays on two different ships and a single source.

The towed array receiver consisted of 64 triplets. A 2 s long HFM pulse with 800 Hz bandwidth was transmitted every 90 s. The data was beamformed, matched filtered, normalised (Kaiser window of 1000 m width), thresholded, clustered [3], and finally tracked using the CMRE tracker DMHT [4].

The data set includes both the monostatic and bistatic tracks, but only the bistatic data set was used, since only bistatic tracks were generated on the pipeline. The data totals 300 tracks, whereof five are on the pipeline. Unfortunately, this is not sufficient to fully demonstrate the method. We have therefore only included a training data set.

**RESULTS**

The proposed method was applied on the data set presented in the previous section, see Fig. 2. Some adjustments were made to the algorithm to take into account the geographical separation between the source and receiver. The necessary maths is
straightforward and therefore not included here. Table 1 lists the parameters tuned for the demonstrated data set. Note that a second, independent data set should have been used as a test data set to validate this choice of parameters, but no such data set was available.

The classification was successful when employing the proposed method with known pipeline positions. All the tracks along the pipeline were correctly classified as pipeline tracks, while the remaining tracks were classified as non-pipeline tracks.

In the case of unknown pipeline positions, one of the pipeline tracks is classified as a non-pipeline, while five of the non-pipelines are classified as pipelines, which amounts to a false alarm rate of 2% and a probability of classification of 67%. The latter number is not statistical significant considering the sparsity of the data. Observe also that due to the requirement in minimum track length, there are less tracks in total; three on the pipeline and 229 tracks with different origins. The pipeline track that the algorithm failed to classify correctly is very short (ten updates, whereof only five had actual detections, while the remaining five were coasts). A stricter requirement on the length of tracks would be to require at least ten track updates with actual detections. Such a requirement results in no false classifications of neither pipeline nor non-pipeline tracks, but also results in a reduction of the data set to only a single pipeline track and 148 non-pipeline tracks.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>$\sigma_T$</td>
<td>160 m</td>
</tr>
<tr>
<td>$\sigma_a$</td>
<td>3.6°</td>
</tr>
<tr>
<td>$\sigma_u$</td>
<td>2 m/s</td>
</tr>
<tr>
<td>$\sigma_\theta$</td>
<td>6.4°</td>
</tr>
<tr>
<td>$T$</td>
<td>1</td>
</tr>
</tbody>
</table>

Table 1: Parameter values used in Eq. (ref{cost}).

CONCLUSION

An algorithm for classification of pipeline tracks has been proposed. The method was tested on a data set from the Norwegian Trench. The proposed method successfully classified all the pipeline tracks without any false alarms when the positions of the pipelines were known. An extended method, where the positions of the pipelines are assumed unknown, was also tested. This method classified 67% of the pipeline tracks correctly, and had a false alarm rate of 2%. All the false alarms and missed classifications occurred on tracks with fewer than ten updates with actual detections. This indicates that the method is more robust for longer tracks.
Fig. 2 Results from the classification procedure on the CEX02 data set. Black lines indicate pipeline positions. The blue line is the path of the sonar vessel. Gray lines are tracks classified as non-pipelines, while red lines are tracks classified as pipelines.

The data set used was not large enough to fully test and demonstrate the method. Ideally, the method should be employed on a training data set in order to determine the set of parameters governing the algorithm, before validating it using an independent data set. Due to the sparsity of the data the validation procedure was omitted from this study.

ACKNOWLEDGEMENTS

We would like to thank all the partners in the NAT III programme. We would also like to thank CMRE for sharing their excellent tracker, DMHT, with us.

REFERENCES

ACOUSTIC OBSTACLE DETECTION FOR SAFE AUV SURFACING

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\textbf{Abstract:} We propose an automatic sea surface object detection from forward looking sonar images. The considered sea surface obstacles are man-made objects: buoys, boats, ships (motorboats or sailboats). Their acoustic signature varies according to their type and state (fixed or moving).

The proposed detection scheme is hierarchical in order to manage the various target signatures. The first step consists in detecting stationary self noise from ships. In case of detection, the strong-intensity strip corresponding to the ship direction is removed to avoid ship noise disturbance during other target detection processes. The next step consists in detecting the other types of obstacles. It is based on an adaptive CFAR (Constant False Alarm Rate) thresholding. The final step consists in analyzing the area around every detected position in order to state that this latter is a reliable obstacle and not a wake signature. Promising results are obtained using real data collected at sea with various objects and scenarios.

\textbf{Keywords:} surface obstacle, detection, obstacle avoidance
1. INTRODUCTION

Obstacle avoidance sonar systems are usually used in a forward looking mode to detect and avoid obstacles either floating in the water column or lying on the sea bottom [1, 2]. Automatic detection and tracking of sea surface obstacles using a forward looking sonar (FLS) is undoubtedly an interesting application that may help for safe submarine and autonomous underwater vehicle (AUV) surfacing. Indeed, in the last decades, many accidents have occurred. A famous one was the collision between the Ehime Maru and the USS Greeneville in 2001.

Surface object detection and classification methods are widely studied in the literature but most of these methods are based on passive noise detection for harbor surveillance purpose [3, 4]. Only few recent methods are proposed for automatic ship detection using active sonar images [5, 6]. However, a single ship is considered at once, with the presence of self noise and wake.

In this paper, we propose an automatic method for the detection of multiple surface objects. The proposed method can detect both fixed and moving vessels with or without self noise and wake.

2. SURFACE OBJECT DETECTION

The considered sea surface obstacles are man-made objects: buoys, boats (motorboats or sailboats), containers, etc. According to the object type and state (fixed or moving), the surface acoustic signature can be:

- Strong-intensity beam(s) in the sonar image (or radial strip(s) in Cartesian representation of sonar data) due to the stationary self noise from ships. This kind of signature indicates the ship direction. It is the strongest among all surface object signatures and the easiest to detect (Fig.1);
- A high contrasted intensity feature in case of noise-free objects like buoys, sailboats and not moving ships such as fishing vessels (see Fig.2);
- Some high intensity lines due to the wake behind a moving vehicle (see Fig.2). The wake echo level strongly depends on parameters like the sensor distance to the ship, the aspect angle and the centre frequency of the transmitted pulse [6].

The proposed detection method consists of three steps. We first try to detect the strongest signature that is the ship stationary self noise. In case of detection, the strong intensity beams corresponding to the ship direction are removed to prevent disturbance from ship noise to the other detection processes. The second step consists in detecting the two other types of signature (object echoes or wake). The third and final step consists in analyzing the area around every detected position in order to state if it corresponds to a reliable obstacle or to wake only.
2.1 SHIP SELF NOISE DETECTION

The detection of the stationary self noise of a ship is often based on the evaluation of the mean energy level along each image beam. The beam of the highest energy is selected to give ship direction [5]. As we assume neither the existence nor the uniqueness of vessel, we search for the more energetic beam(s) with respect to their neighborhoods. In addition, instead of using the average of the intensity beam as an indicator of ship self noise presence, we use the median value of each image beam that is more suitable to the multiple target case. Indeed, the average of the high intensity of a given beam may correspond to several targets without stationary noise and not to a single ship self noise. Formally, the indices of the bearing measurements associated to ship self noise in the image at ping $k$ are given by:

$$\left\{ \eta_k^j \right\} = \left\{ \left| \text{med}(j) > \text{Th} \right| \text{mean} \left( \text{med} \left( j - N : j + N \setminus \{ j - N_g : j + N_g \} \right) \right) \right\}$$

where $\text{med}(j)$ is the median of the image beam $I_k(\cdot, j)$ and $\text{med} \left( j - N : j + N \setminus \{ j - N_g : j + N_g \} \right)$ the set of medians of its $2 \times N$ neighboring beams.
except the $2 \times N_g$ closest ones. $Th$ is a preselected threshold, $mean$ stands for the average operator and $N_g$ is set according to the angular sampling $T_s$ and to beam spacing $\sigma_\delta$ as follows:

$$N_g = \text{round} \left( 3 \times \frac{\sigma_\delta}{T_s} \right)$$

According to the fact that ship echoes are usually strong, ship range detection consists in finding the highest intensity pixels in the detected beams. Because of the time varying gain (TVG) or the automatic gain amplifier usually applied to sonar images, ship noise level increases with range and the target could be masked especially if it occurs at far range. To tackle this problem, as in [5], we first normalize every detected image beam by the median value of a sliding beam segment:

$$\tilde{I}_k(m,n) = \frac{I_k(m,n)}{\text{med} \{ I_k(m-N_b,n), ..., I_k(m+N_b,n) \} }, \quad n \in \tilde{\Phi}_k$$

where, $I_k(m,n) = I_k[m\Delta d, n\Delta \delta]$ is the value of the image $I_k$ at polar coordinates $(m\Delta d, n\Delta \delta)$, $\Delta d$ is the range sampling rate, $\Delta \delta$ is the bearing sampling rate and $2 \times N_b + 1$ is the length of the sliding segment.

And then the first ten strongest peaks of the normalized detected beams $\tilde{I}_k(:,n), n \in \tilde{\Phi}_k$ are kept. However, if the ship wake is also visible and the wake end detected, the number of detections is reduced to only one as explained in section 2.3.

### 2.2 NOISE FREE SURFACE OBSTACLE DETECTION

To our knowledge, there is no work dealing with the detection of ships without self noise in active sonar images. In SAR imagery, the topic of ship detection has been extensively studied. The most powerful detector used in SAR imagery for this purpose is the CFAR (Constant False Alarm Rate) detector. This detector uses an adaptive threshold computed for each image cell (or pixel) by taking into account the clutter power estimated from a set of reference cells surrounding the cell under investigation as depicted in Fig. 3.

![Fig.3 Adaptive CFAR detector: $P_{fa}$ is the probability of false alarm and $S_{CFAR}$, the detection threshold.](image)

Guard cells are the nearest neighbors of the cell under investigation and are not used in the local clutter estimation in order to ensure that pixels belonging to an extended target
cannot be included in the clutter statistics estimation. The CFAR threshold $S_{\text{CFAR}}$ is set so as to ensure the expected false alarm rate $P_{fa}$.

Adaptive radar detection for several clutter probability density functions (Gaussian, Weibull, K-law, Pearson, etc) have been widely studied in the literature [7, 8]. Here, we deal with amplitude sonar images and the distribution of pixel levels is usually modeled by a Rayleigh density [9]: $f(x) = \frac{2x}{B} \exp\left(-\frac{x^2}{B^2}\right)$. We assume that clutter samples are independent and identically distributed and we estimate the parameter $B$ according to the maximum likelihood method [8]:

$$\hat{B} = \left(\frac{1}{M} \sum_{i=1}^{M} x_i^2\right)^{1/2},$$

where $M$ is the number of reference cells. The CFAR detector is then based on the following threshold [8]:

$$S_{\text{CFAR}} = \left[ \left( P_{fa} \frac{1}{M} - 1 \right) \sum_{i=1}^{M} x_i^2 \right]^{1/2} \quad (1)$$

### 2.3 WAKE DETECTION

Ship wakes are made of dense air bubble clouds created by the propeller [6]. They extend on hundred meters and last for some minutes according to ship propeller, hull design, ship speed and maneuver [6].

Wake pixels have in general high levels and are often detected by the above CFAR detector (section 2.2). The detection of wake pixels increases the amount of false alarms and can lead to inaccurate ship position estimation.

In order to discriminate between the detection associated to the reliable ship position and the detection related to wake only, we first search for wake and we assume that ship position is given by a wake endpoint with the highest echo level. However, in our real data, we notice that for some cases, ship echo is the wake endpoint with the lowest energy especially for short and strong wakes. Hence, we decide to detect both wake ends.

Wake end detection consists in analyzing the area around every detected position within a set of oriented strips as depicted in Fig.4.

![Fig.4 Wake detection procedure for a given detected position (in yellow), in red are presented four oriented strips of 15 m length and 20 degrees width: on the left, polar plot of the sonar data and on the right, Cartesian plot.](image)
We consider that a given detection is associated to a wake if the number of detections in one of its associated strip of length $L_1$ for a given orientation is over a preselected threshold $N_{W'}$. This last is set heuristically according to the probability of false alarms. Depending on the value of $N_{W'}$, several detections inside the wake and especially near every extremity usually satisfy this condition. The endpoint is the detection whose associated strip has the largest amount of detections.

3. RESULTS

The proposed approach is assessed on real FLS images with several fixed buoys with different sizes laid on the sea-surface and moving ferries and sailboats.

The experiments were conducted by ENSTA Bretagne in Brest harbor, in June 2013. The acoustic data are recorded with the RESON 8101 FLS mounted on the ENSTA Bretagne survey boat at roughly 2 meters below the sea surface. The sonar looks forward but intercepts the sea surface (see Fig. 5). The recorded data covers a 60 degrees sector in the broadside direction. The sonar central frequency is equal to 240 kHz and its bandwidth is about 15 kHz. The beamwidth is 1.5 degree and the range resolution is about 5cm.

Fig. 5: Sonar configuration.

In Fig. 6 we can see detection results of a passenger boat quitting the harbor: the ferry direction is showed by white lines, candidate measurements for its distance are plotted in red and in magenta is the final range deduced from wake detection.

Fig.6 A ferry detection: the ferry self noise detection (white lines), candidate range detections (red) and final range deduced from wake detection (magenta); on the left, polar plot and on the right, Cartesian plot.
In Fig. 7, we can see results of buoys detection with different sizes and of two concrete blocks detection near the pier for $P_{fa} = 10^{-5}$. Three of these objects are seen in the photo (Fig. 7 (a)): a large buoy and two concrete blocks. Two smaller buoys are behind the large one (in white circles in Fig. 7 (b)).

All known surface objects are detected. However, as the detections resulting from the proposed method are point (or pixel) measurements and as most obstacles have a large extent, several detections are obtained for each obstacle. Some isolated false alarms are related to sea clutter. Their amount depends on the $P_{fa}$ value.

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**Fig. 7** Fixed object detection: (a) photo of the surface objects, (b) Cartesian plot of sonar data: circled in red are the objects seen on the photo and in white the smaller buoys behind the large buoy, (c) detection point results are in red.

---

In Fig. 8, we show the detection result of a zodiac and a sailboat both with a wake. We notice that even for low $P_{fa}$ values ($P_{fa} = 10^{-6}$ in this case), wake cells are detected. White points correspond to the ends of wakes. Here, for each vessel, the two wake extremities are kept. However thanks to the sequence of images, the wake extremity associated to the ship position could be detected through the analysis of vessel trajectories.

---

**Fig. 8** A zodiac and a sailboat detection: from left to right: photo of the zodiac and the sailboat, Cartesian plot of sonar data and detection results: detected points in red and detected wake end points in white.

---

**4. CONCLUSION**

Automatic detection of sea surface obstacles in looking forward sonar images is a challenging issue dedicated to safe submarine and autonomous underwater vehicle (AUV) surfacing. The proposed algorithm can detect multiple obstacles at once with various
signatures, i.e. fixed and moving vessels with or without self noise and wake. Promising results have been obtained with real data gathered at sea. At this time, detections only consist of points (pixels) that can relate to the same large obstacle. Hence, further improvements concern detected points clustering. A region growing based segmentation method using these detected pixels as seed points is under investigation. And so, obstacle segmentation will provide a better target description (size, shape, and so on).

At last, the proposed detection method can be used together with a Kalman filter within a surface obstacle tracking framework.

5. ACKNOWLEDGEMENTS

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Abstract: MIMO sonar systems can offer great capabilities for area surveillance especially in very shallow water with heavy cluttered environment. We present here a MIMO simulator which can compute synthetic raw data for any transmitter/receiver pair in multipath and cluttered environment. Synthetic moving targets such as boats or AUVs can also be introduced into the environment. For the harbour surveillance problem we are interested in tracking all moving objects in a particular area. So far the tracking filter of choice for multistatic systems has been the MHT (Multiple Hypothesis Tracker). The reason behind this choice is its capability to propagate track identities at each iteration. The MHT is an extension of a mono object tracker to a multi object problem and therefore suffers from a number of drawbacks: the number of targets should be known and the birth or death of new tracks are based on heuristics. A fine ad hoc parameter tuning is then required and there is a lack of adaptivity in this process. To overcome those restrictions we will be using the HISP (Hypothesised multi-object filter for Independent Stochastic Population) filter recently developed. The HISP filter relies on a generalisation of the concept of point process that integrates a representation of distinguishability. As a consequence, this filter deals directly with the multi-object estimation problem, while maintaining track identities through time without using heuristics. While filters track the objects after processing in the digital domain, we show as well in this paper that we can adapt acoustical time reversal techniques to track an underwater target directly with the MIMO system. We will show that the proposed modified DORT technique matches the prediction / data update steps of a tracking filter.

Keywords: MIMO sonar systems, tracking, time reversal.
1. **INTRODUCTION**

Multiple Input Multiple Output sonar systems have raised a lot of interest during the recent years mainly in the ASW community. Multi-static sonars overcome monostatic sonar systems in target localisation and detection performances [1]. CMRE in particular developed a deployable low frequency multi-static sonar system called DEMUS. The DEMUS hardware consists of one source and three receiver buoys and can be denominated as a SIMO (Single Input Multiple Output) system. A lot of the efforts were focussed on the data fusion and the target tracking problems. Several trackers including centralised and decentralised MHT (Multi-Hypothesis Tracker) [2] or TBD (Track Before Detect) trackers [3] have been developed and applied to the DEMUS datasets.

In this paper we present a full 3D MIMO simulator which can compute synthetic raw data for any transmitter/receiver pair in multipath and cluttered environment. Synthetic mid-water targets can also be added to the environment. MIMO image formation will be discussed and MIMO autofocus techniques will be demonstrated. We show in particular that the depth of a mid water target can be estimated with great accuracy. The principles of the MHT filters will be discussed and the HISP filter will be presented. The HISP deals directly with the multi-object estimation problem, while maintaining track identities through time without using heuristics. Finally we will show that large MIMO systems offer an ideal platform for time reversal techniques. We will present in particular an unfocussed time reversal mirror algorithm capable of tracking automatically moving targets.

2. **MIMO SIMULATOR**

2.1. **Seabed interface**

To model the seabed interface we generate 2D fractional Brownian motion using the Incremental Fourier Synthesis Method developed by Kaplan and Kuo [4]. The main idea is to model the 1st and 2nd order increments $I_x$, $I_y$ and $I_2$. $I_2$ for example is given by:

$$I_2(m_x, m_y) = B(m_x + 1, m_y + 1) + B(m_x, m_y) - B(m_x, m_y + 1) - B(m_x + 1, m_y)$$

(1)

where $B$ is the 2D fBm. Those 1st and 2nd order increments can be computed thanks to their FFTs. The 2nd order increment FFT is given by:

$$S_2(\omega_x, \omega_y) = \frac{32 \sqrt{\pi} \sin^2(\omega_x/2) \sin^2(\omega_y/2) \Gamma(2H + 1) \sin(\pi H)}{\sqrt{\omega_x^2 + \omega_y^2}}$$

(2)

where $H$ is the Hurst parameter. Figure 1 displays an example of 2D fractional Brownian surface generated using this technique.

2.2. **Bistatic reverberation level**

The bistatic scattering strength is computed using the model developed by Williams and Jackson [5]:

$$S_b(\theta_s, \phi_s, \theta_t) = 10 \log[\sigma_{br}(\theta_s, \phi_s, \theta_t) + \sigma_{bv}(\theta_s, \phi_s, \theta_t)]$$

(3)

where $\sigma_{br} = [\sigma_{kr}^n + \sigma_{pr}^n]^{1/n}$ is the bistatic roughness scattering which includes the Kirchhoff approximation and the perturbation approximation. $\sigma_{br}$ is the sediment bistatic volume scattering. $S_b$ depends on the bistatic geometry as well as the sediment physical properties. Figure 2 displays the bistatic scattering strength for a Tx/Rx pair situated 141m apart and both at 7.5m from the seafloor. The $S_b$ is computed for
two different sediment types (coarse sand and sandy mud) for the same fBm interface. There is around 10dB difference is the $S_b$ for the two sediments which can plays a role in the detection/tracking process. We will consider these two sediment types later on.

![Fig. 1: Example of 2D fBm with $H = 0.8$ (fractal dimension = 2.2)](image)

Fig. 1: Example of 2D fBm with $H = 0.8$ (fractal dimension = 2.2)

Fig. 2: Bistatic scattering strength relative to one Tx located at [0m,100m] and a Rx located at [100m,0m] for (a) a coarse sand sediment type and (b) a sandy mud sediment type.

2.3. Propagation

Sound propagation in shallow water can become extremely complex. Because we are modelling harbour environment we assume a constant sound speed through the water column. To model the multipath we are using the mirror theorem. In conjunction with a constant sound speed ray tracing techniques are used to compute the different propagation paths. The simulations done in this paper consider a maximum of three bounces. The reason behind this choice is that the coherent MIMO processing done on the next section suppresses greatly incoherent echoes.

To synthesise time echo a random scatterer point cloud including random position and random intensity is generated for each cell in the seabed. Note that once the point cloud is generated, it can be saved for other simulations with the same configuration.

In our case we want to synthesise time echo from $400 \times 600$ cells $\times 20$ scatterers per cell $\times 100$ MIMO pairs which represents around half a billion paths to compute (direct paths only). Brute force computation using MatLab on a standard laptop requires around 2 months of computation. Hopefully a handful of tricks can reduce drastically this time. One of them is to use sparsity with the the circular
convolution properties of the DFT. The main tool to propagate a signal is free water is the well known FFT property: \( f(t - t_0) \leftrightarrow e^{-i\omega t_0} F(\omega) \). If we consider the echo related to one cell, this echo is extremely sparse over a 600m range signal. The idea is to compute the propagated signal over a much smaller window. Figure 3 draws the outlines of the algorithm: the full scene is divided into range bands, on Fig. 3(a) each colour band represents a 10m range division. The echoes relative to each band are computed independently on a small window of 20m (cf. figure 3(b)). The echoes are then recombine to give the full range bistatic response as seen in figure 3(c). Using those techniques greatly reduces the computation time from 2 months to around 10 hours.

\[ \text{Fig. 3: (a) of the observed scene in 10m range band. (b) Individual range band echo contribution. (c) Full echo response recomposition.} \]

2.4. MIMO imaging and autofocus

In order to image the output of the MIMO system we will use the multi-static back-projection algorithm which is a variant of the bistatic back-projection algorithm developed by the SAR community. Further details can be found in [6]. Using the back-projection algorithm the Synthetic Aperture Sonar (SAS) image is computed by integrating the echo signal along a parabola. In the bistatic case the integration is done along ellipses. For the multi-static scenario the continuous integration is replaced by a finite sum in which each term corresponds to one transmitter/receiver pair contribution. Figure 4 displays a synthetic aperture MIMO image: the background is a fractal coarse sand seafloor, a mid-water
target is present at the location [200m, 150m].

As it has been mentioned before synthetic aperture MIMO imaging shares a lot of features with standard SAS imaging. In particular the image is projected onto a plane or a bathymetry estimate. The image of a mid water target will then appear unfocused for this particular projection. By moving the projection plane through the water column the MIMO target image will focus at its actual depth. Using simple autofocus algorithm it is then possible to estimate the depth of the target even if the MIMO system is coplanar. For a mid water target at 400m range in a 15m depth environment it is possible to estimate its depth with 10 to 50 cm accuracy. Figure 5 displays the autofocus results and the estimated target depth compared with the ground truth.

![Fig. 5: Autofocus algorithm results based on maximising the scattering response: ground truth (white curve) and estimated depth (green curve).](image)

3. HARBOUR SURVEILLANCE SCENARIO USING MIMO SONAR SYSTEMS

![Fig. 6: (a) Harbour scenario. (b) geometry of the MIMO simulation. (c) Colour coded detections: (light blue) static bottom object, (dark blue) false alarm, (yellow) fish, (orange) boat, (red) AUV.](image)
4. MULTIPLE OBJECT TRACKING

After the derivation, in the 1960’s, of the first principled single-object filter, known as the Kalman filter, the problem of tracking multiple targets in a cluttered environment rapidly arose. As a consequence, gradually sophisticated methods for handling the complexity of data association have been introduced. These methods can be seen as bottom-up approaches, as they build up multiple target tracker from the Kalman Filter. One of the most successful of these methods is the MHT [7], which principles and limitations are summarised in Section 4.1. Since early 2000, another class of methods, which we will describe as “top-down”, have been introduced. These methods are presented in Section 4.2.

4.1. The MHT filter

The MHT, for Multiple Hypothesis Tracking, is a multi-target tracker that handles data association in a probabilistic way. It can be seen as one of the most sophisticated bottom-up tracker, building on the idea behind techniques such as the GNN, for Global Nearest Neighbour, or JPDA, for Joint Probabilistic Data Association, while incorporating the concept of hypothesis. However, the MHT also have shortcomings, (a) it is much more computationally demanding than the GNN or JPDA, and is known to be intractable for complicated target tracking problems, and (b) it inherits from the ad-hoc management of birth and death found in any bottom-up approach.

4.2. The PHD and HISP filters

In 2003, the PHD filter [8], for Probability Hypothesis Density, has been introduced in order to address the limitations of the MHT. It can be seen as one of the first top-down approaches to the problem of multiple target tracking. The PHD filter, and other similar filters, are based on the principled modelling of the multiplicity which is inherent to target tracking, and allow for the integration of birth and death of targets in a probabilistic and consistent way. The issue of computational complexity is also addressed by assuming that tracks are not distinguishable, so that they can be represented by a single distribution over the state space. However, track identities are lost as a consequence, and additional algorithms have to be used in order to recover the estimated state of each track. The impact of this limitation is strengthen by the use of multiple dynamical models for the propagation of each track, or when classification is required.

Recently, a new multi-target tracking algorithm called the HISP filter, for Hypothesised filter for Independent Stochastic Populations, has been introduced [9, 10]. The HISP filter presents the same advantages as the PHD filter but maintains track identities. This is made possible through the introduction of distinguishability into the multi-target representation. As a consequence, any single-object filter can be used within this multi-target framework, including classification, as demonstrated below.

4.3. Results

The output of a Gaussian Mixture implementation of the HISP filter, or GM-HISP, is pictured in Figure 7 with two different types of seabed: Figure 7b for coarse sand and Figure 7c for muddy sand. These figures show that the HISP filter managed to separate the fishes from the other targets. This is made possible by estimating two different multi-target populations with two different dynamical models. In order to distinguish the static targets from the boats and the UAV, a Sequential Monte Carlo implementation of the HISP filter, or SMC-HISP, would be required, as dynamical models excluding small velocities are non-Gaussian. More specifically, the coarse sand scenario 7b has more false alarms than the muddy
sand scenario 7c. As a result, the estimation is made more difficult, e.g. the estimated positions of the fishes are not as consistent as the one given for the muddy sand scenario, the latter being closer to the ground truth.

5. ACOUSTICAL TRACKER

Prada et al. in [11] described the iterative time reversal process for a static scene. The MIMO problem formulation can written as:

\[ \mathbf{R}(\omega) = \mathbf{K}(\omega)\mathbf{E}(\omega) \]  \hspace{1cm} (4)

where \( \mathbf{E}(\omega) \) is the column vector of the FFT of the transmit signals, \( \mathbf{R}(\omega) \) is the column vector of the FFT of the received signals and \( \mathbf{K}(\omega) \) the channel matrix. Given a received signal \( \mathbf{R}_n(\omega) \), the next output signals is given by: \( \mathbf{E}_{n+1}(\omega) = \mathbf{K}^*(\omega)\mathbf{E}_n(\omega) \). With this formulation and collocated Tx and Rx, the \( 2n^{th} \) input signals is:

\[ \mathbf{E}_{2n}(\omega) = [\mathbf{K}^*(\omega)\mathbf{K}(\omega)]^{n}\mathbf{E}_0(\omega) \]  \hspace{1cm} (5)

Prada shows the convergence of the \( [\mathbf{K}^*(\omega)\mathbf{K}(\omega)]^n \) operator to the brightest scattering point of the scene. Effectively the MIMO array focus the sound to this scattering point. For a dynamic scene \( \mathbf{K} = \mathbf{K}(\omega,t) \) varies with time. We note \( \mathbf{K}_n(\omega) \) the channel matrix at time step \( n \). We can now write: \( \mathbf{R}_n = \mathbf{K}_n\mathbf{E}_n \). In order to track an underwater target in motion we propose to defocus the input signal \( \mathbf{E}_{n+1} \) accordingly to the maximum speed of the target. \( \mathbf{E}_{n+1} \) becomes \( \mathbf{E}_{n+1} = \mathbf{G}\mathbf{K}_n^*\mathbf{E}_n^* \). Equation 5 then becomes

\[ \mathbf{E}_{2N} = \left[ \prod_{2n=2}^{2N} \mathbf{G}\mathbf{K}_2n_{-1}\mathbf{G}^*\mathbf{K}_{2n-2} \right] \mathbf{E}_0 \]  \hspace{1cm} (6)

It is interesting to note that the iterative defocussed time reversal process it equivalent to the general approach taken by digital tracking filters. Tracking algorithms proceed in two steps:

\[ p_k(X_k|Z^{(k)}) \rightarrow p_{k+1|k}(X_{k+1}|Z^{(k)}) \rightarrow p_{k+1}(X_{k+1}|Z(k+1)) \]  \hspace{1cm} (7)

The first step is a prediction step and is equivalent to the defocus operator \( \mathbf{G} \). The second step is the data update is equivalent to the channel matrix operator \( \mathbf{K}_n \).
6. CONCLUSIONS

In this paper a full 3D realistic MIMO sonar simulator was presented. We showed the value of large MIMO sonar systems for underwater surveillance. In particular we studied the problem of harbour surveillance and underwater object tracking. The traditional MHT and PHD filter approaches were discussed and results using the HISP filter were presented. Finally we proposed a time reversal approach to tracking using defocused output signals.

7. ACKNOWLEDGEMENT

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QUANTIFYING THE COMPLEXITY IN SONAR IMAGES FOR MCM PERFORMANCE ESTIMATION

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Abstract: Seafloor characteristics like roughness and clutter density affect the complexity in sonar images, significantly influencing the achievable performance for detection and classification of bottom targets, e.g. mines. The introduction of autonomous underwater vehicles (AUV) into mine countermeasures (MCM) operations has thus created a need to measure such properties in imagery from high-resolution side-looking sonar (real or synthetic aperture) using a consistent metric. In this paper, we present our approach for quantifying the complexity in sonar images with a continuous, multi-scale measure based on local variances of wavelet coefficients obtained by the two-dimensional maximal overlap discrete wavelet transform (MODWT). Moreover, we determine the presence of anisotropy, i.e. image textures with a clear directionality like sand ripples, through a combination of ratios from wavelet variances and rotated integral images. The proposed method is applied to synthetic aperture sonar (SAS) images, where promising results are achieved both in terms of validity and computational time. We see image complexity as an essential parameter for MCM performance estimation. Further it can also be used to improve the performance of automatic target recognition (ATR) by adapting the processing to the given environment.

Keywords: Synthetic aperture sonar (SAS), mine counter measures (MCM) performance, maximal overlap discrete wavelet transform (MODWT), complexity, anisotropy
1. INTRODUCTION

The seafloor characteristics significantly influence the achievable performance for Mine Counter Measures (MCM) operations where Autonomous Underwater Vehicles (AUVs) equipped with high-resolution, side-looking sonar are used to search for mines. Seafloor roughness, clutter and reverberation level gradients induce sonar image intensity patterns that may resemble or interfere with mine responses, thus decreasing the probability for mine detection and classification, increasing the number of false alarms, or both. This performance degradation will be large for most automatic target recognition (ATR) algorithms and somewhat smaller for a human operator analyzing the sonar data.

One attempt to account for this has been to segment the seafloor into regions of different types based on image texture descriptions, and establish the corresponding performance for each type, see e.g. [1,2]. The problem is that unsupervised classification methods yield unlabeled types lacking physical interpretation, while results from supervised methods depend heavily on the training set. Also, many seafloors do not consist of well-defined texture regions, as features like roughness and clutter can be present to a varying degree. This favors texture characterization over classification.

In this paper we thus follow the approach of [3,4], which described wavelet related methods to provide continuous measures for the properties seafloor complexity and anisotropy. The complexity indicates how difficult it is to classify a mine given the local seafloor texture while the anisotropy indicates the presence of a dominant texture orientation (e.g., sand ripples). This approach enables performance evaluation of previously unseen seafloor types. A related work estimated complexity from image entropy [5].

We use the term image complexity instead of seafloor complexity as in [3,4]. This is to emphasize that the complexity is calculated from the sonar image, which is a function of more than the seafloor features. Measurement geometry, given by aspect and grazing angles, affects the sonar image significantly, particularly for oriented seafloor textures like ripples. Fluctuations in the water column like internal waves can introduce signal variations even for smooth seafloors [6]. Sonar conditions are also important. As an example, complexity will be low for sonar images heavily contaminated by sea surface multipath even for textured seafloors. This demonstrates that image quality is also a critical mine hunting performance parameter [7].

Our work on image complexity is part of a larger FFI effort to develop an MCM performance assessment tool for mine hunting with AUVs. The current prototype focuses on the mine detection and classification part of the full MCM performance. It uses local, in-situ values for image complexity and image quality, in addition to the estimated fraction of undetectable mines due to, e.g., burial. Mine threat is also an input parameter, as it is more difficult to classify modern stealth mines than traditional cylinder mines. Additionally, if complexity and anisotropy measures are produced in the AUV, they can be used by the vehicle’s autonomy system to adapt the mission plan. In the case of strong anisotropy, survey lines can be adapted to minimize image complexity. Cross lines can be added to increase total performance in high complex areas. Finally, complexity and anisotropy measures can be used as input to ATR systems, which can then apply parameters and algorithms optimized to the actual image textures.

The main technical contribution of this paper is that complexity is calculated over multiple scales with focus on mine-size. We make use of the maximal overlap discrete wavelet transform (MODWT) [8-10] which enables us to decompose the variance of a process over different spatial scales. Further, we incorporate coherence-based signal to noise ratio (SNR) of the underlying sonar image [7] in a final smoothing step.
2. THEORY

2.1. Two-dimensional MODWT

Here we give a brief introduction of the MODWT to the extent necessary to understand our proposed method. For more details, the reader is referred to, e.g. [8,10]. MODWT is a shift-invariant and non-subsampled version of the more known Discrete Wavelet Transform (DWT). There are no restrictions on the length of the underlying process and the resulting coefficients will be of the same size as the input, independent of the chosen scale. These properties make it well suited for our subsequent scale-based analysis of complexity in sonar images.

The \( j \)th level wavelet and scaling filters of length \( L_j \) will be denoted as \( \{ h_{j,l} \}_{l=1}^{L_j} \) and \( \{ g_{j,l} \} \). These are differencing and averaging filters respectively, here chosen from the Daubechies class [8,11] of which its simplest unit level filter with length \( L_1=2 \) is known as the Haar filter.

For a two-dimensional process \( \{ X_{u,v} \} \) on a regular lattice, the resulting wavelet coefficients depend on application of the scaling or wavelet filter in the horizontal and vertical direction. We follow [9] and obtain the two-dimensional wavelet-wavelet (\( \text{ww} \)), scaling-wavelet (\( \text{sw} \)) and wavelet-scaling (\( \text{ws} \)) coefficients for levels \( j \) and \( j' \) by a tensor product of the filters. As described in [10], \( \text{ww} \) coefficients mostly capture diagonal, \( \text{sw} \) vertical and \( \text{ws} \) horizontal structure from the underlying process \( \{ X_{u,v} \} \). Levels \( j \) and \( j' \) are associated to scales \( \tau_j = 2^{j-1} \) and \( \tau_{j'} = 2^{j'-1} \) for wavelet filters \( \{ h_{j,l} \} \) and \( \{ h_{j',l} \} \), respectively to \( 2\tau_j \) and \( 2\tau_{j'} \) for scaling filters \( \{ g_{j,l} \} \) and \( \{ g_{j',l} \} \). The coefficients can be calculated efficiently with fast Fourier transform operations for each desired scale.

The wavelet variance for levels \( j, j' \) is defined as \( \nu_{K,j,j'}^2 = \text{var}\{K_{j,j'}\} \), where \( K \) here symbolizes either of the coefficients (i.e., \( \text{ww}, \text{sw} \) or \( \text{ws} \)). Given a realization of \( \{ X_{u,v} \} \) of size \( N \times M \), an unbiased estimator for the wavelet variance is shown in [9]:

\[

\nu_{K,j,j'}^2 = \frac{1}{N_jM_j} \sum_{u=-L_j-1}^{N_j-1} \sum_{v=-L_j-1}^{M_j-1} K_{j,j'}^2(u,v), \text{ where } N_j = N - L_j + 1 \text{ and } M_j = M - L_j + 1. \tag{1}

\]

For the remains of the paper we choose the same horizontal and vertical scales, i.e. \( j=j' \).

2.2. Local multi-scale wavelet variances

We consider wavelet variances for each scale locally within overlapping image boxes of size \( N_b \times M_b \). For this we first calculate the two-dimensional wavelet coefficients on the entire image followed by a shift to correct for the non-zero phase of the filters such that events in the coefficients align with the events in \( \{ X_{u,v} \} \) [8,12]. Then, we estimate the local variances on the non-boundary affected inner regions by

\[

\nu_{K,j,j'}^2(u,v) = \frac{1}{N_bM_b} \sum_{u=-[N_b/2]-1}^{[N_b/2]-1} \sum_{v=-[M_b/2]-1}^{[M_b/2]-1} \tilde{K}_{j,j}(u_b,v_b), \tag{2}

\]

where \( \tilde{K} \) denotes either of the shifted coefficients. For the boundary affected regions, some adaptations are needed; e.g., zero-padding or using only the reduced number of coefficients available.
We define a measure $0 \leq \xi \leq 1$ at location $(u,v)$ as the weighted sum of the local wavelet variances over several scales $i=j_{\text{min}}$, ..., $j_{\text{max}}$, with a stronger weight $w_{j_m}$ on mine-size scale with index $j_m$:

$$\xi(u,v) = \frac{\sum_j \hat{v}_{ij}^2(u,v) w_j}{\sum_i w_i},$$

where $\hat{v}_{ij}^2(u,v) = \max(\hat{v}_{sw,j}^2(u,v) + \hat{v}_{sw,j}^2(u,v) + \hat{v}_{sw,j}^2(u,v), t) / t$.  

As explained with the tri-diagonal representation in [9], the variance for a wide range of processes can be decomposed into an infinite series of $ww$-, $sw$- and $ws$-variances over all scales; i.e.,

$$\text{var}(X_{u,v}) = \sum_{j=1}^{\infty} \nu_{sw,j}^2 + \sum_{j=1}^{\infty} \nu_{sw,j}^2 + \sum_{j=1}^{\infty} \nu_{ws,j}^2.$$  

We are using this to put a threshold $t$ on $\hat{v}_{ij}^2(u,v)$ in Eq.(4). Normalizing the image by the standard deviation of a worst possible example of clutter, threshold $t$ determines that only a certain percentage of the overall variance shall be contained in the decomposed local variance per scale.

The multi-scale variance measure $\xi$ will be smoothed with a mean filter weighted by the coherence-based SNR of the underlying sonar image [7]. This will remedy the problem of locally low $\xi$ values for the shadows in mine-size rock fields.

### 2.3. Image complexity and anisotropy

Following [10], we make use of the fact that under isotropy (i.e., direction independence) variances of $sw$ and $ws$ are equal; i.e., $\nu_{sw,j}^2 = \nu_{sw,j}^2$. This is equivalent to $\log(\nu_{sw,j}^2 / \nu_{ws,j}^2) = 0$. The more the log-ratio differs from zero, the stronger is the presence of a preferred direction, hence anisotropy. However, the above condition is also fulfilled if the anisotropy is $45^\circ$. Thus we need a measure that takes this into account. Our suggestion is to consider an extra pair of filters that are $45^\circ$ rotated from the original filters. This is feasible in the case of Haar wavelets as they can be also calculated by integral images [13] to achieve similar results (see Fig. 1). Bearing the above isotropy relation in mind, we construct an anisotropy measure that uses both ratios of estimated local $sw$- and $ws$- variances from Eq. (2) and corresponding $45^\circ$ rotated coefficient ratios obtained through rotated integral images [14] with width and height matching the scale of the $sw$- and $ws$- coefficients. Simulations of random fields with differently oriented major anisotropy axis indicate stability of this approach.

Anisotropy $A(u,v)$ is similarly weighted as the multi-scale measure in Eq. (3), but less scales will be considered since we want to focus on anisotropy caused by sand ripples, which typically are smaller than mine-size. On higher scales the anisotropy of the seafloor often is caused by larger geophysical processes such as currents. Further, we limit the degree of anisotropy by introducing upper and lower thresholds on the ratios to make it

\[2\tau_j - \tau_j \]

\[2\tau_j + \tau_j \]

**Fig. 1:** Illustration of scaling-wavelet and wavelet-scaling Haar coefficients calculation and their $45^\circ$ rotated integral image version. Each sub-block has length $2\tau_j$ on the long side and $\tau_j$ on the short one. The average from one sub-block is deducted from the corresponding average of the other.
more robust against outliers. As for $\xi$, after combining the result for several scales we perform a SNR weighted final smoothing step.

A region of structured, directional response like sand ripples can lead to higher values of the multi-scale wavelet variance measure $\xi$, which we perceive as not optimal. Regions with clear directionality are easier to analyze than randomly cluttered ones with similar variance and thus should be rated less complex. To account for this, we construct a complexity measure $C$ based on $\xi$ incorporating anisotropy $A$ to reduce the effect of sand ripple regions in $\xi$. Hence we define complexity $C$ at location $(u,v)$ with weight $0 \leq \gamma \leq 1$ as

$$C(u,v) = \xi(u,v)(1 - \gamma A(u,v)).$$

(6)

Complexity $C$ achieves its lowest values for uncluttered, smooth sediment seafloors and the highest for densely cluttered seafloors. Border areas between seabed types with significantly different reflectivity will be clearly enhanced.

3. DATA EXAMPLES

The measures described above are demonstrated on two data sets collected with HUGIN AUVs in 2009 outside Horten (Norway) and in 2013 during MANEX’13 [15] at the south side of Elba island (Italy). Both vehicles were equipped with HISAS 1030 sonar. For the analysis in this paper, we use magnitude sonar images on logarithmic scale. The grid sizes in the across and along track direction were both around 3.8cm. Hence we can use the same horizontal and vertical scales; i.e. $j=j'$. We define $j_m=5$ as mine-size scale index, which corresponds to horizontal, vertical and diagonal structure in areas of size around $1.2 \times 1.2m$.

Fig. 2 displays results for a mud seafloor outside Horten, illustrating the scale-based decomposition in form of $\nu_j^2$ from Eq. (4) for $j=1,\ldots,7$ using the Haar wavelet and the resulting complexity image $\hat{C}(u,v)$ according to Eq. (6) with $\gamma = 2/3$. The SAS image (upper left) reveals considerable clutter in the central lower part, a small ship wreck on the left side, scattered debris and rocks across the image as well as low scale structure caused by biological bottom activity.

The two lowest scales describe differences of neighboring pixels or pixel pairs. They are affected mostly by speckle noise, which explains the rather high values for $j=1$ and $j=2$. This is less relevant for mine hunting, hence those scales should be excluded for our further analysis. As expected, the ship wreck is best visible on higher scales, while the large clutter area becomes less present with increasing scales with index $j > 3$, but negligible only from $j=7$ on. Therefore this area gives high image complexity values when focusing on levels $j=\{3,4,5,6\}$ with strongest weight for $j_m=5$.

The influence of anisotropy in form of sand ripples is shown in Fig. 3 for an area outside the island of Elba. The multi-scale wavelet variances produce medium high values for the sand ripple field in the upper middle of the image. Our anisotropy measure recognizes this area such that the sand ripple influence is reduced in the final complexity image.

FFI is currently implementing a tool for MCM performance estimation where image complexity is an integral part. Although being a property of the sonar image and not just the seafloor, we have found it useful to display image complexity also as geographical maps. We create those maps by geo-referencing and fusing all relevant complexity images from a mission, keeping the maximum pixel value for overlapping grid cells. The operator can then use the map to quickly establish how the complexity varies over the whole area, or alternatively examine single survey line results.
Fig. 4 shows a SAS mosaic and the corresponding complexity map for a much larger area outside Elba than the scene in Fig. 3. On the lower left part, there are several rock formations yielding high complexity values. In the central part we see some degree of small scale structure, but to a rather low extent. The large bright region in the upper part of the SAS mosaic is seagrass (posidonia oceanica), while the dark areas are exposed sediments. The dark horizontal stripes correspond to the nadir region without sonar coverage. Closer to the shore (upper map edge), the seagrass becomes sparse and the sharp transitions between the bright seagrass and the dark sediments create high complexity values. The areas fully covered by seagrass are fairly smooth in the sonar mosaic and give low complexity values.

Fig. 2: Port-side SAS image section of seafloor outside Horten, Norway (top left) and resulting complexity (bottom right) as well as the estimated summed wavelet variances $\psi^2_{jk}(u,v)$ for scale indices $j=1,...,7$. In all images the displayed area covers 75-212m across track and 112m along track.

Fig. 3: Images from left to right: SAS (starboard), multi-scale wavelet variance $\xi(u,v)$, anisotropy $A(u,v)$ and complexity $C(u,v)$. The displayed area covers 19-114m across track and 113m along track. The area of high anisotropy values is caused by sand ripples in the SAS image.

Fig. 4: Images from left to right: SAS (starboard), multi-scale wavelet variance $\xi(u,v)$, anisotropy $A(u,v)$ and complexity $C(u,v)$. The displayed area covers 19-114m across track and 113m along track. The area of high anisotropy values is caused by sand ripples in the SAS image.
One could argue that these areas should be flagged as highly complex because mine hunting will be difficult. This is, however, due to mines becoming buried in the vegetation and not due to large intensity variations. Our MCM concept has other parameters to address this issue; i.e., the fraction undetectable mines.

4. SUMMARY

In this paper we have described an approach to assess image complexity for the use in MCM performance estimation. We prefer to work on image complexity rather than seafloor complexity because MCM analysis is based on the sonar image contents. In addition to seafloor properties, these will be affected by measurement geometry, water column characteristics and sonar conditions.

Following other publications, we sustained the terms complexity and anisotropy, where the first is more important for us. Image complexity is a topic of growing interest due to its versatility in use for MCM, ATR, change detection or (adaptive) mission planning. The novelty in this paper is that we describe it in terms of a multi-scale measure based on capturing intensity variations through wavelet variances using MODWT. Though we illustrated examples using the Haar wavelet, the framework is in place for using also longer filters from the Daubechies wavelet class. Further, we include SNR in the final smoothing process, a measure that can be obtained by interferometric sonar coherence. Our complexity is computed efficiently using fast Fourier transform in the wavelet coefficient calculation. Due to the structure of wavelets, most of the information can be retained even when downsampling the input, thus speeding up the calculations even more; e.g., if needed to run in delayed real-time.

Our approach shows promising results on data collected by HUGIN 1000 AUVs equipped with HISAS 1030 sonar. The described method is flexible enough to handle also data from other sensors and/or vehicles. Care must be taken only when the image resolution differs for both directions. Here wavelet coefficients with $j \neq j'$ [16] will be useful, alternatively scaling of the input in one direction to achieve approximately the resolution of the other.

Fig. 4: SAS mosaic of a mission on the south side of Elba during MANEX’13 (left) as well as the resulting complexity map (right) including black lines to display the trajectory. The red rectangle in the lower part of the SAS mosaic indicates the area displayed in Fig. 3.
REFERENCES


TOWARDS AUTOMATIC TARGET RECOGNITION IN LOW-FREQUENCY SUB-SEDIMENT SONAR IMAGERY

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Abstract: Detection of unexploded ordnance is challenging in the underwater environment, particularly when object burial occurs. A capability to detect buried targets has been demonstrated previously using TNO’s MUD low frequency sediment-penetrating sonar and other similar sonars. However, the high clutter rates encountered in practice have the potential to impose severe operational limitations in absence of a robust capability to distinguish targets from clutter. To this end, we are taking the initial steps towards development of an automatic target recognition algorithm for detecting targets and suppressing clutter in low-frequency sub-sediment sonar imagery. The initial implementation presented in this paper uses a previously developed wavelet shrinkage algorithm to suppress the background reverberation, followed by automatic thresholding and segmentation to isolate individual seafloor objects for subsequent extraction of their acoustic signatures. We show preliminary detection results from the MUD-2011 data set.

Keywords: Synthetic aperture sonar, buried targets, automatic target recognition

1. INTRODUCTION

In an underwater environment, the detection of unexploded ordnance (UXO) is difficult when the objects are partially or fully buried. Sine burial frequently occurs, either in conditions with soft seafloors (silt or mud) or due to sediment transport, methods need to be developed to aid the detection of UXO in such conditions. Broadband low-frequency sonars are a promising technology for sub-sediment imaging and detection of buried objects [1],[2],[3]. The low frequencies enable penetration into the seafloor sediment, while broadband signals facilitate potential classification of objects based on the multi-
aspect acoustic colour. TNO’s hull-mounted side-looking low-frequency synthetic aperture sonar (referred to as the MUD sonar) has been developed for this purpose [1].

UXO detection is challenging with the MUD sonar due to high levels of reverberation, the presence of clutter, and low target echo amplitudes. To develop an effective target detector, a two-stage approach is proposed:

1. A contact detector for selecting objects of interest, including UXO and clutter.
2. A classifier which extracts features from the contacts with the objective to distinguish UXO from the clutter.

This paper focuses on the first stage detector mentioned above. For this purpose, dedicated image processing has been developed and applied to experimental data. The image processing is summarised in Section 2, and consists of synthetic aperture processing, followed by target/background separation. Section 3 discusses the contact detector. The results on data acquired by the MUD system are discussed in Section 4 and conclusions are drawn in Section 5.

2. MUD SONAR AND PROCESSING CHAIN

The MUD sub-sediment imaging sonar is a side-looking system with flexible tilt angle. It has a 16-element vertical array, enabling the signal to reverberation ratio to be improved by the suppression of multipath reverberation. It has an accurate navigation system comprised of RTK-GPS and INS, and a horizontal array to aid the synthetic aperture processing. The system is capable of operating in the frequency range between 1 kHz and 30 kHz [1].

2.1. Multipath Suppression and SAS Processing

In shallow water, strong multipath interference can mask the echoes from targets and corrupts their acoustic signatures, particularly at longer ranges. This adversely affects detection and classification performance. Vertical beamsteering is applied for the mitigation of the multipath interference as described in [4].

Synthetic aperture sonar (SAS) uses coherent processing of the echo data to attain high resolution in the along-track direction. This is especially important for low frequency sonars, which typically have poor resolution due to their wide beams. The applied time domain back-projection SAS processing includes motion compensation, and is described in detail in [4].

2.2. Target / Background Separation by Incoherent Wavelet Shrinkage

Effective detection of targets in background reverberation noise requires a robust method, which is able to discern even weakly scattering objects in a highly reverberant seafloor. A coherence metric can be used to separate the targets from the background. In [5], a coherence metric was derived to determine the similarity of wavelet coefficients between independent looks, i.e. different images of the same scene with statistically
independent noise realizations. It is assumed that a high coherence corresponds with the reverberation-free measurements of targets, while the low coherence contributions are assumed to correspond to reverberation (background). By weighting the image according to this coherence, a separation can be made between targets and background. A thorough description of this method can be found in [5].

3. TARGET AND CLUTTER DETECTOR

The detection of buried targets in sonar images is difficult due to the presence of clutter and reverberation. This is illustrated in Figure 1a, where the ground truth positions of deployed objects are overlaid on a SAS image from the MUD sonar. Here, we present an automated approach to obtain contacts corresponding to both targets of interest and clutter. This is a first basic step towards automatic target detection. Once these contacts are obtained, more detailed information can be derived using dedicated processing, for example based on features of the multi-aspect acoustic colour [4].

The operation of the contact detector is described in Figure 2 and consists of the following steps:

1. Apply a fixed threshold to the SAS image leading to a binary image.
2. Apply morphological opening and closing operations to obtain pixel clusters.
3. Generate a list of contacts from the pixel clusters.

To assess the performance of this basic detector, the known ground truth positions of targets are associated with the detected contact positions, taking into account an assumed maximum distance between a contact and the ground-truth position. The purpose of this step is to estimate the probability of detection (i.e. whether contacts are generated corresponding to deployed targets) and the number of false alarms (i.e., the remaining non-associated contacts). Since the number of false alarms in an image is a non-normalised number, depending on the size of the image, it can be normalised by the area. Here, the normalisation area is chosen to be a square swath-width (50 m × 50 m). A receiver operating characteristic (ROC) curve is obtained by plotting the proportion of detected targets (true positive rate) versus the normalised number of false alarms (false positive rate).

After sonar data processing, a SAS image is obtained as illustrated in Figure 1a. This image is naturally used as input for the detection process (shown in the flow diagram in Figure 2 as a dashed line) to provide a baseline performance estimate. However, because the image contains a clearly visible background and clutter contacts, it is anticipated that the thresholding will not be effective and will result in many false alarms. Therefore, a second pre-processing step is considered, whereby a coherence filtering operation (incoherent wavelet shrinkage) is used to separate the image into coherent and incoherent components; these are assumed to correspond with the objects and background reverberation, respectively. The coherent part is shown in Figure 1b.

In the next section, the performance of the detector is evaluated when it is directly applied to the SAS image, and when the incoherent wavelet shrinkage technique is included in the processing chain to remove the background reverberation.
Figure 1  (a) SAS image and ground truth target positions, marked with a square; (b) the same image, after wavelet shrinkage, showing the coherent part.

Figure 2  Flow diagram of the image processing and target detection chain.
Figure 3  The binary contact clusters (after the morphological opening and closing step) for both the SAS image (a) and the contacts image (c), and the resulting detections overlayed on the SAS image (b) and the coherent image (d). Ground truth positions are marked with a cyan square, while contact detections are marked with a green circle. The colour scale is identical in the images and is normalised between 0 and 1.
4. RESULTS

The performance of the detection chain is influenced by the choice of a few parameters (shown on the right-hand side of Figure 2). Here, the structuring elements are fixed with a circular element of 3 pixels in diameter for the opening and a circular element with 7 pixels in diameter for the closing. The maximum distance for the association between a ground truth target position and a detected contact is chosen to be 2 m. The only remaining parameter is the detection threshold, which affects the trade-off between detection performance and false alarm rate. For normalised images, this is varied between 0 (all pixels) to 1 (no pixels). Default parameters are used in the wavelet shrinkage algorithm [5].

Example results using a detection threshold of 0.05 are shown in Figure 3. The top image (a) shows the thresholded SAS image after the morphological operations. As can be seen, clutter from the background is still present, and these contacts contribute to the false alarms. In (b) the resulting detections are overlaid on top of the SAS image. The large number of false alarms confirms the prediction that a second pre-processing step is necessary. The same procedure is repeated for the coherent image, where the background clutter has been suppressed. The resulting binary image (c) and the overlay on top of the SAS image (d) show a much better result. The example reveals that the automated target detection chain applied directly to the SAS image has a 100% probability of target detection, but at the cost of 276 false alarms per square swath width (i.e. 2500 m²). When applied to the coherent image, the probability of detection is 94% (16 out of 17), but with a much lower false alarm rate of 15 false alarms per square swath width.

The procedure can be repeated for a range of detection thresholds to yield a ROC curve. The ROC curves are shown in Figure 4 for two separate runs, each containing the same 17 targets, with and without the background removal. As can be seen, the number of false alarms is significantly reduced using the proposed method, although at the cost of some mis-detections.

5. CONCLUSIONS AND DISCUSSION

We have described and demonstrated a basic contact detector for a low-frequency sediment-penetrating sonar. The detector is sensitive to UXO and clutter and will provide the first stage in a future automatic detector. The performance of the proposed detector was demonstrated with ROC curves and compared with the performance of a basic threshold detector.

We have shown that a substantial improvement in false alarm suppression is achieved by applying a wavelet shrinkage pre-processing step before applying the detection threshold. Although the false alarm suppression is improved by this approach at lower sensitivities (roughly 1 order-of-magnitude better), the probability of detection is affected at higher sensitivities, attaining only 90-95% detection probability at the highest sensitivities compared to 100% for the simple detector (at the cost of a very high false alarm rate). This suggests several improvements that will be investigated in future work:
1. Optimisation of the detector parameters, including settings of the wavelet shrinkage algorithm, morphological operators, and clustering; and 2. using a combination of results from the processed and non-processed images. Moreover, using the contact detector, we intend to extract signatures from many UXO and clutter contacts to establish a robust feature set for UXO detection.

6. ACKNOWLEDGEMENTS

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A GPU SONAR SIMULATOR FOR AUTOMATIC TARGET RECOGNITION

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Abstract: Template matching is a common technique for automatic classification of objects in synthetic aperture sonar (SAS) images. The principle is to isolate an image segment containing the object of interest, correlate it with a set of template images, and assign it to the class of the template yielding the highest correlation coefficient. The challenge is to come up with a representative set of template images covering the relevant configurations of object and seabed.

We target this challenge with a sonar simulator that first takes as input a seabed model derived from the real sonar image. Then it places a 3D object model on the seabed, renders the scene, and adds the resulting image to the template set. For every object type, position, alignment and material, the procedure is repeated, and a correlation coefficient computed. The best performance is obtained when these parameters are estimated from the sonar image as part of the classification process. The simulator is therefore written in OpenGL and OpenCL and runs on graphics processing units (GPUs). The result is a fast performing and portable on-the-fly template generator which can adapt to the characteristics of the current scene.

Keywords: Sonar, SAS, simulator, template matching, OpenGL, OpenCL

1. INTRODUCTION

Automatic target recognition (ATR) is an important component in autonomous underwater vehicles (AUVs), as it allows the vehicle to adapt its mission plan to e.g. revisit detected objects for closer examination. One way to classify these objects is by using template matching. The principle involved is to isolate an image segment containing the object of interest, correlate it with a set of template images of the relevant object classes, and assign it to the class of the template yielding the highest correlation coefficient. The challenge is to come up with a representative set of template images covering the actual configurations of object and seabed. The
alternative of using a static, predetermined template library has a computational complexity that is exponential with the number of parameters. This limits its use to only a few parameters being coarsely sampled, ultimately yielding inaccurate results [1].

We have developed a synthetic aperture sonar (SAS) simulator that avoids this problem by running sufficiently fast to create templates adapted to the actual scene as part of the classification process. This is achieved with the aid of the massive computing power in graphics processing units (GPUs) and optimized software libraries for scene rendering. The simulator loads a 3D model of the seafloor and an object class model into OpenGL, where emitted sound waves are modeled with a light source placed at the sonar transmit location. We assume rough, isotropic surfaces reflecting sound energy equally in all directions. This can be modeled with a Lambertian scattering model [2, 3], where the intensity of the backscattered sound only depends on the incidence angle of the transmitted signal onto the model surface. When rendering OpenGL is set up to produce an optical 2D intensity image and depth map that reveals the distance from each pixel to the propagation axis. For maximum flexibility this data is finally combined with OpenCL to produce the image templates.

Various simulators for high frequency, side-looking sonar imagery have been published, e.g. [4, 5]. Most implementations have however prioritized accurate acoustic modeling at the expense of execution speed. One exception is the SIGMAS+ simulator in [6, 7] which, similar to our approach, takes advantage of parallel processing at GPUs and uses OpenGL to render a 2D image from a 3D model, under the assumption of Lambertian scattering. However, SIGMAS+ is still aimed more towards realistically looking images, including effects like ambient noise, which is irrelevant for our purpose of template generation. Also, their simulator sums images obtained from multiple OpenGL rendering passes to create the sonar image, while we render once and post-process this result with OpenCL to tailor the sonar image with more flexibility.
2. METHODS

Creating a 2D view of an arbitrary complex 3D scene is a non-trivial matter. This is why we decided to use the core OpenGL pipeline to do this for us. OpenGL is a popular, well-matured and multi-platform application programming interface (API) for rendering 2D and 3D vector graphics. It relieves us from the intricacies of projecting vertices, faces and textures defined in a 3D space onto a suitable 2D image plane. This section explains how we set up OpenGL for this task, and then proceeds to describe how we post process the OpenGL images with OpenCL to form the final sonar image.

2.1 Setting up the scene

Before we can render anything with OpenGL we first have to set up the scene. This involves loading the models and then placing them where we want them with the proper orientation.

Our models are stored in regular 3D model files. The simulator loads these with the Open Asset Import library (assimp). This is a portable open source library that supports loading a wide range of 3D model formats in a uniform manner. Given a model file it outputs the model as a node tree where node contains data such as vertices, facets and textures, all formatted in an OpenGL friendly way.

The next step is to scale, orient and position the objects in the scene, and finally project this scene onto an image plane. This is achieved by applying a set of transformations to each vertex in the model. In OpenGL, the set of transformations is usually formalized like this:

\[
\begin{bmatrix}
   x \\
   y \\
   0 \\
1
\end{bmatrix}_{image} = P \cdot V \cdot M \cdot \begin{bmatrix}
   x \\
   y \\
   z \\
1
\end{bmatrix}_{model},
\]

where \( \begin{bmatrix} \cdot \end{bmatrix}_{model} \) is the loaded model, \( \begin{bmatrix} \cdot \end{bmatrix}_{image} \) is the output image, and \( M, V \) and \( P \) are transformation matrices. The first transform we apply is the model matrix \( M \), which scales and rotates
the model in its local coordinate system. Then we map this model into world coordinates using the view matrix $V$. This involves translating the model to its respectful place in the world, and then moving it into the view of the camera. Finally the relevant part of the scene is projected onto the image plane with a projection matrix $P$.

2.2 OpenGL rendering

To produce the sonar templates we assume a rough, isotropic surface that reflects energy equally in all directions. This permits us to use a Lambertian scattering model where the backscatter intensity depends only on the incidence angle [8]. It does not consider observation angle or sound frequency, but for the purpose of creating templates this is not needed.

The rendered image will appear as if we placed a window at the sonar and looked through it in the direction of the image. This window is resized to make sure that the template image perfectly fills it.

2.3 OpenCL post-processing

The “camera image” rendered with OpenGL is not in along-track and cross-track coordinates as we want it to be. It does, however, tell us what parts of the scene that are visible from the sonar. OpenGL can also produce a depth map that reveals the distance to each of image pixels. This information can be converted to a ranged sonar-like image by simply adding up all the intensity values that share the same depth for each range line. We perform this computation in OpenCL, which allow general purpose programming on GPUs and can interoperate with OpenGL quite nicely. This way we keep all the calculations on the GPU.

3. RESULTS & DISCUSSION

Our GPU-based simulator is very fast. On a computer with a six-core Intel Core i7-3930K and a Radeon HD7970 it computes almost 1000 templates per second, each being 1 megapixel large. This makes it possible to tailor the templates very closely imaged objects. In contrast, in the standard approach a template library must be created beforehand from a limited set of parameter values. Hence, it may happen that no template in the library is a close fit of the object in the image even though the object actually is a target of interest.

This performance was demonstrated on an image from the HISAS 1030 synthetic aperture sonar mounted on the HUGIN AUV. Out of the objects in the scene we selected a 2.6 m long cylinder that was partly buried in the sea sediments (Fig. 3a). The length, immersion depth and aspect angle of this cylinder were estimated from the SAS image using our adaptive template matching approach [1]. Then these parameters were used by the simulator to create an adaptive template (Fig. 3b), which is an almost perfect match of the cylinder. We also created standard templates for a regular template matching approach. In this case, we assumed that the cylinder mine had a generic length of 2 m (like a MP80 or a Murena mine) and was proud on the seafloor. Moreover, templates were created for every 10° degree of the aspect angle. We believe these assumptions to be typical for a template library for cylinder mines.

The standard and adaptive templates were then matched to the HISAS image. This is illustrated in Fig. 4a for a standard template and in Fig. 4b for an adaptive template. These images were created by segmenting the image and the templates into highlight, shadow, and background regions. The regions from a template were then laid on top of the regions of the image to illustrate how well these regions matched. Note how the adaptive template obtained
(a) SAS image of a cylinder with 2.6 m length and 0.53 m radius.
(b) Cylinder simulation with parameters estimated from SAS image: Length 2.6 m, burial depth 0.263 m, aspect angle 105°.

Fig. 3: A SAS image of a cylinder and a template simulation adapted to it.

(a) Best template out of a predefined set. Inaccurate cylinder length, alignment and immersion depth. The segmented shadow and highlight regions are misaligned.
(b) Adapted template. Template simulated with adapted parameters. Good fit in both highlight and shadow segments.

Segmentation overlay color description:
- Template highlight and image highlight
- Template highlight only
- Image highlight only
- Background pixels
- Template shadow and image shadow
- Template shadow only
- Image shadow only
- Template shadow and image highlight

Fig. 4: Segmented simulation image overlaid the segmented SAS image.
a much closer fit to image than the standard template. This was also reflected in the correlation scores (which were created with the method described in [1]) that were 0.613 for the standard template and 0.813 for the adaptive template. Hence, the standard approach was less likely to classify the cylinder in the image correctly as the standard templates were not created specifically for the target.

4. CONCLUSION

One way to automatically classify objects in SAS images is to compare the imaged objects with a predefined set of templates. However, this is suboptimal as it is infeasible to create accurate templates for all the relevant configurations of seabeds and objects. To solve this we have implemented a SAS simulator that creates the templates in delayed real-time based on parameters estimated from the current scene. In our studies this improves the match between the SAS objects and their corresponding template significantly in most cases.

To obtain the delayed real-time performance we implemented the simulator on a GPU. These devices have a theoretical peak performance that is typically an order of magnitude higher than CPUs in a comparable price range. We found this potential to be effectively utilized with the well matured and highly optimized OpenGL graphics processing API. Our simulator use this framework for most of the scene processing.

A final post-processing step is performed in OpenCL, which allow general purpose GPU programming. It inter-operates well with OpenGL and adds a lot of flexibility to the simulation process. This is valuable as the simulator is still being actively developed and will likely see new features that can not be easily implemented in OpenGL alone.

REFERENCES

EFFICIENT SUPERELLIPSE FITTING BASED CONTOUR EXTRACTION FOR MINE-LIKE SHAPE RECOGNITION

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\textbf{Abstract:} For the classification of mine-like objects in sonar images, the contour of its shadow and the features derived from it play an important role. Therefore, the extracted contour needs to be of high quality, i.e. undisturbed by noise and artefacts. In this paper superellipse fitting is presented as a way to improve the extracted contour after segmentation. A new linearization of the fitting task is introduced to speed up the computation time. Moreover, linear tapering and circular bending are considered in the context of fitting more complex contours to the shadow of mine-like objects. In the experimental investigations several different methods of fitting are compared with regard to the resulting fitting error and computation speed. Results suggest that the linearization indeed speeds up the computation by a factor of 2 up to 4 without any loss of the fitting accuracy.

\textbf{Keywords:} superellipse fitting, contour extraction, linearization
1. Introduction

Autonomous underwater vehicles (AUVs) are one of the key technologies in the 21st century. The scope for such high-end sensor platforms are both, civil and military applications. AUVs are usually equipped with high resolution imaging sonar systems. To fulfil the demand of an as autonomous as possible behaviour, the AUV has to be capable of correctly interpreting the gathered sonar data. Among other things, the shadow of a recognized object in a sonar image is used to derive its class. Thus, the contour of the shadow is extracted via segmentation, and besides others, several contour based features are computed. In doing so, the quality of the contour features depends heavily on the quality of the extracted contour. The topic of this paper is to present a way to improve an extracted contour via superellipse fitting and new approaches to do the actual fitting. Superellipses, or sometimes called Lamé-curves, are a compact and efficient way to describe a variety of shapes in a closed expression. By altering the squareness of the superellipse function rectangles, rhomboids and ellipses can be generated. Up to date, superellipse fitting has been used in several fields of computer vision. Besides other applications, they were even used in the context of detection and classification of mine-like objects [1],[2]. What separates this work from the afore mentioned is a new linearization to speed up the computation, and the use of two deformation transformations, linear tapering and circular bending, to enhance the modelling fitting.

2. Different fitting approaches

Superellipses

Superellipses are defined implicitly as

\[ \left( \frac{x_s}{a} \right)^\varepsilon + \left( \frac{y_s}{b} \right)^\varepsilon = 1. \]

Figure 1 illustrates the effect of different values for the squareness \( \varepsilon \). Besides the implicit expression the parametric equation for a superellipse is

\[ x_s = a (\cos \theta)^\varepsilon \]

Figure 1: Different superellipses for constant \( a \) and \( b \) and varying \( \varepsilon \). From the outermost to the inner superellipse \( \varepsilon = 0.5, 1, 2 \) and 5.
with the parametric value $\vartheta \in [-\pi; \pi]$. To avoid imaginary solutions only $\vartheta \in [0; \pi/2]$ is considered. Consequently, every data point has to be mirrored into the first quadrant. The superellipse fitting will be done by minimizing a fitting error. Thus, a suitable fitting error has to be defined. The straightforward approach would be to measure the distance for each data point between this point and the superellipse point with the shortest Euclidean distance to the corresponding data point. Unfortunately, the computational cost for this approach is quite high, so that various distance approximations were introduced. The authors of [3] compared nine of them. Unfortunately, the result was that there is no obvious best error measure and the performance depended on the level of noise. In view of the application of enhancing contours extracted out of sonar images, the authors of this paper decided to use an error measure, which worked reasonable well under noise. The idea is to define the corresponding point on the superellipse as the intersection point of the line between the origin of the superellipse and the data point and the superellipse itself [4]. Thus,

$$\begin{align*}
y_s &= b \left( \sin \vartheta \right)^\varepsilon \\
\text{with } \vartheta \in [-\pi; \pi].
\end{align*}$$

This can be inserted into (1) to calculate for any given data point $(x_{d,i}, y_{d,i})$ and parameters $a, b$ and $\varepsilon$, the corresponding point on the superellipse. With this, the fitting error measure is

$$\hat{q}(a, b, \varepsilon) = \sum_{i=1}^{N} \left( x_{d,i} - x_s \right)^2 + \left( y_{d,i} - y_s \right)^2.$$  

In the remaining part of this chapter different approaches are presented to demonstrate how the fitting can be actually done.

### Three parameter problem

This is the straightforward version. Hereby, the three parameters must be found that minimize the fitting error in (3). The minimization problem is

$$(a^*, b^*, \varepsilon^*) = \arg\min_{a,b,\varepsilon} \hat{q}(a, b, \varepsilon).$$

Any iterative solver can be chosen to solve this optimization problem.

### Eight parameter problem

Here, instead of three parameters for all data points, every quadrant gets its own three parameters [1]. To ensure a continuous graph of the resulting approximation, the semi axes are linked so that

- $a_1 = a_4$, $a_2 = a_3$, $b_1 = b_2$, $b_3 = b_4$,

with the indices indicating the quadrant. This constraint reduces the number of parameters from twelve to eight.

### 4 x 3 parameter problem

Basically, the 4x3 parameter problem equals the eight parameter version. However, applying this approach the three parameter problem is solved first for each quadrant independently, without any coupling between the semi axes. Afterwards, the appropriate semi axes are
averaged to obtain the final eight parameters [1].

**Linearization**

In order to partly linearize the minimization problem the ratio between \(a\) and \(b\) for every quadrant is iteratively fixed. Thus, \(\theta\) is within an iterative step only dependent on \(\varepsilon\) and the fitting error can be written as

\[
q(\theta, \varepsilon) = (X - H(\varepsilon)\theta)^T((X - H(\varepsilon)\theta)),
\]

with \(\theta = (a, b)^T\), \(x = (x^T_a, y^T_a)^T\) and

\[
H(\varepsilon) = \begin{pmatrix}
c(\varepsilon) & 0 \\
0 & s(\varepsilon)
\end{pmatrix},
\]

\(c(\varepsilon) = (\cos(\theta_1)^\varepsilon, \cos(\theta_2)^\varepsilon, \ldots, \cos(\theta_N)^\varepsilon)^T\), \(s(\varepsilon) = (\sin(\theta_1)^\varepsilon, \sin(\theta_2)^\varepsilon, \ldots, \sin(\theta_N)^\varepsilon)^T\). Minimizing (4) with respect to \(\theta\) leads to

\[
\nabla_{\theta} q = 0 \Rightarrow H(\varepsilon)^TH(\varepsilon)\hat{\theta} = H(\varepsilon)^T x.
\]

Formally the solution is

\[
\hat{\theta} = (H(\varepsilon)^TH(\varepsilon))^{-1}H(\varepsilon)^T x. \tag{5}
\]

If (5) is inserted in (4) the fitting error becomes

\[
q_p(\varepsilon) = q(\hat{\theta}, \varepsilon) = X^T(Id - P(\varepsilon))x
\]

\[
P(\varepsilon) = H(\varepsilon)(H(\varepsilon)^TH(\varepsilon))^{-1}H(\varepsilon)^T.
\]

The algorithm to calculate the fitting parameters is

1. Initial guess for \(c = a/b\).
2. Determine \(\theta_i\) via (2).
3. Solve \(\varepsilon^* = \arg\min_{\varepsilon} q_p(\varepsilon)\).
4. Insert \(\varepsilon^*\) in (5) to receive \(\theta^*\)
5. \((a^*, b^*) = \theta^*\).
6. Update \(c = a^*/b^*\).
7. If change in \(a^*, b^*\) and \(\varepsilon^*\) is small break, else go to step 2.

Depending on the initial guess for \(c\), the algorithm needs only very few iteration steps. For the following experiments, we allowed only two iteration steps. This linearized approach can be transferred to the 8 parameter problem and the 4 x 3 parameter problem quite easily.

**Linear Tapering and circular bending**

Linear tapering and circular bending are common deformations applied to superellipses or superquadrics in 3D fitting approaches [6], [7]. Formally, the tapering and bending transformations in 2D are defined as

\[
B(x, y, k) = \left( x + \frac{1}{k} - \cos(ky) \left( \frac{1}{k} - |x| \right) - |x|, \sin(ky) \left( \frac{1}{k} - 1 \right) \right)
\]

\[
T(x, y, t) = \left( \frac{t}{b} y + 1 \right) x, y
\]

Figure 2 shows the different effects of both deformations. As the tapering and bending operations are not commutative, it has to be decided which one to apply first. The authors of [6] state that when a combination of deformations is used, the more shape preserving one should be made first. Since the extracted contours have frequently a straight shape, the bending
operation was chosen first. Both deformations are used in conjunction with the three and the eight parameter problems. Therefore, the non-linear solver has to solve a problem with two additional variables each.

3. Experimental results

In a first experiment each fitting approach is tested on 86 contour examples of real detections. The origin of the coordinate system is for each contour in its centre. The y-axis is aligned with the main orientation of the contour. This is done via a principle component analysis. The Nelder-Mead algorithm [4] was used as iterative solver, whenever required. Starting Values are gathered be ellipse fitting. Figure 4 shows the fitting results made by the different approaches for one example. As expected, the three parameter approach provides the poorest approximation. The eight parameter, or with tapering and bending ten parameter approach, seems to give the best result. All other approaches lie between these two. Overall, the difference between the eight parameter and the 3x4 parameter variants is small. This impression remains true for all other examples not shown here. It is interesting to note that the linearization in the 3x4 case enhances the error, whereas in the eight parameter case the opposite is true. The first row of Table 1 shows the fitting error over all examples averaged over the number of points of all contours combined. It is ordered from the worst performing to the best performing approach. These values confirm the impression from Figure 4.
An important factor, besides the accuracy, is the computational effort a fitting algorithm yields. Table 2 shows the average computation time for each approach. By dividing the eight parameter problem into four three parameter problems, the computation time decreased by approximately the factor four. Linearization furthermore decreased the computation time so that the linearized 4x3 parameter problem is approximately 20 times faster than the 8 parameter problem. Tapering and bending, however, increased the computational effort by the factor of 2 to 3.

In a second experiment, since the data set of real examples is limited, a model contour with added noise is used. The model contour and one corrupted example can be seen in Figure 3. The noise is generated by a random walk of a scaled Gaussian pseudorandom variable with mean 0 and standard deviation 1. To smooth the noise a digital filter is applied. This time the real contour is known, thus, after the different fitting algorithms are employed, in the end the difference between the model contour and the approximation is calculated. The average result for 100 runs can be found in the second row of Table 1. The order from the worst to the best approximation stays mostly the same.

<table>
<thead>
<tr>
<th>method</th>
<th>SV</th>
<th>3PP</th>
<th>TB3PP</th>
<th>4x3PP</th>
<th>T4x3PP</th>
<th>L4x3PP</th>
<th>L8PP</th>
<th>8PP</th>
<th>TB8PP</th>
</tr>
</thead>
<tbody>
<tr>
<td>Real</td>
<td>0.4034</td>
<td>0.4015</td>
<td>0.3948</td>
<td>0.3356</td>
<td>0.3354</td>
<td>0.3029</td>
<td>0.2954</td>
<td>0.2796</td>
<td>0.2708</td>
</tr>
<tr>
<td>Synt.</td>
<td>0.3973</td>
<td>0.3973</td>
<td>0.3953</td>
<td>0.2503</td>
<td>0.2502</td>
<td>0.2470</td>
<td>0.1983</td>
<td>0.1957</td>
<td>0.1927</td>
</tr>
</tbody>
</table>

Table 1: Sum of fitting error of all examples averaged over all data points.
In this paper, new methods of fitting a superellipse onto a given contour were presented and compared with existing approaches. Experiments suggest that with the help of the introduced linearization a significant speed up in the computation time can be achieved, while the resulting approximation of the contour is equally accurate. Adding tapering and bending in the context of contours of mine-like objects improved the approximation, albeit barely. Unfortunately, at the same time the computational effort increased significantly. With this in mind there are two options. If there is no time constraint and the approximation of the contour has to be as good as possible, one should choose the 8 parameter approach with tapering and bending. However, if a time constraint exists, the linearized 3 x 4 approach that nearly achieves the same fitting quality for only a fraction of computational effort should be favoured.

### Literature


IDENTIFYING CONTENT OF LOW PROFILE TARGET IN CLUTTERED ENVIRONMENT USING THE BIOSONAR

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Abstract: In the mine counter measures context image based automatic target recognition has two main limitations: in a cluttered and heavily cluttered environment recognition algorithms observe a drastic increase in the P\textsubscript{Fa} (probability of false alarm) making the output results impractical at the best. The second main limitation is unknown threats such as IEDs (Improvised explosive devices) or simply unknown types of mines. In this paper we present classification results from a trial done in Portland harbour, Weymouth, UK in October 2012. The test targets were 9 identical gas cylinders (65 cm height, 30 cm diameter) filled with three different contents: 3 cylinders were filled with seawater, 3 with sand and 3 with gravel. The targets were placed on two different highly reflective seabed types (mud with broken shells). The estimated SNR of these low profile targets in these difficult environments is negative making them almost undetectable in traditional sidescan images. We demonstrate here the capability of the wideband BioSonar to identify and distinguish between the 3 types of cylinders. We present in situ learning using spiral reacquisition pattern and show that the wideband classification is robust to environment variations.

Keywords: ATR, wideband sonar.
1. INTRODUCTION

The focus in automatic target recognition for MCM has mainly been in image processing. The introduction of SAS (Synthetic Aperture Sonar) has given access to very high resolution sonar images and strengthened the image based approach. Heavily cluttered environments and IEDs, however, are still a significant problem for this approach. In this paper we present MCM trial results using a wideband sonar. We demonstrate the capability of the wideband system to identify and distinguish between identical cylinders with different fills.

2. BIOSONAR

2.1. BioSonar characteristics

The wideband sonar (WBS) system used during these trials was supplied by Hydrason Ltd. The original design for this system was based on the biological sonar used by bottlenose dolphins (tursiops truncatus) and the current incarnation retains many of the original operational parameters. The effective bandwidth is nominally 30 to 130 kHz. The importance of having a wideband sonar system for object recognition has been explored in [1]. The BioSonar was operated in sidescan mode with two complementary transducers, high-frequency (HF) with a centre frequency at around 90 kHz to port side and low-frequency (LF) with a centre frequency at around 60 kHz to starboard.

A variety of pulses were used during the trials. For this paper however we focus our analysis in one longer duration (600 µs) downswept linear FM pulse. The rise and fall periods for this pulse follow half-Gaussian patterns to avoid excessive ringing. The time signal and magnitude spectrum are given in Figure 1.

![Time Domain](image1)

![Frequency Domain](image2)

Fig. 1: Pulse used for the trials. (left) time domain, (right) frequency domain

The beamwidth is wide by conventional sonar standards. The main purpose for a wide beamwidth is to ensure whole object ensonification. The importance of transducer beamwidth for target recognition is discussed from a stochastic point of view in the next section 2.2.

The BioSonar sensor was mounted at the front of Heriot-Watt University’s REMUS-100 autonomous underwater vehicle (AUV) and complemented the standard fit Marine Sonics sidescan unit. A sensor depression angle of 10° for the BioSonar system was used throughout and all missions reported here were run at a nominal altitude of three metres. The echoes were recorded with 16 bits precision at 800 kHz sampling frequency and at around 10 Hz ping repetition rate.
2.2. BioSonar Beamwidth

The beamwidth of the sonar system is a joint function of the transmit and receive beam patterns. For the current BioSonar system two receive settings are available. For the narrower receive setting, the joint -3 dB beamwidth of the system is around 6° at 120 kHz. At the lower frequency end (around 40 kHz), the joint beamwidth of the sonar is around 18°. We can reasonably assume that the beamwidth variation across the frequency band is near linear. For missions following a classic lawnmower pattern, the sonar beamwidth impacts directly on the number of hits per pass. Since the beamwidth is frequency dependent, this also varies with frequency. For views of targets acquired at a range of 15 to 20 m, we can expect to record between 8 and 20 hits per pass depending on the frequency band. The number of hits per pass is an important consideration from a statistical point of view. If we consider the recorded echoes as a stochastic process (this hypothesis is especially valid for seabed returns and cluttered environments), probability dictates that over a one hour mission (around 10^6 pings) a not insignificant number of returns may give responses similar in some way to a target echo we are interested in.

In the following paragraphs we aim to demonstrate from a stochastic point of view the importance of the wide beamwidth of the BioSonar and the importance of the echo consistency between hits. For the sake of the discussion, let us simplify the problem and let us consider that the echo amplitude variable is similar to an unbiased coin (output value: 0 or 1). Consider a sampling interval of 10 cm and suppose that we are looking for a specific target with highlights as shown in figure 2 i.e. with the specific sonar response: 1010000001. The target of interest covers 10 samples which corresponds to a 1 m extent. Note that though this echo model may appear oversimplified, it matches exactly the dynamic one can expect in the feature domain. Over 1 m range data we can reasonably extract around 10 significant features. The reader can refer to [1] for feature extraction in wideband sonar data for object recognition.

In our model the seabed response will follow the random variable described earlier and the target response will be deterministic. In order to compute the false alarm rate we just need to calculate the chance of observing an echo from the seabed matching the specific target: \( \left( \frac{1}{2} \right)^{10} \approx 9.77 \times 10^{-4} \). So, the probability of such an event is relatively low, but considered in the context of an hour-long mission with 10 Hz ping rate and 60 metres range covered by each ping, we can estimate that this particular event will occur on average more than 2000 times per mission. Assuming a transit of 3 knots, an hour-long mission will cover around 0.5 km^2 of terrain. Using the information on only one ping will then result in more than 4000 false alarms per km^2.

Trials data have shown repeatedly that target echoes change slowly over small angular changes during a transit. So we can assume that the target echo will retain some consistency over the number of hits per pass. From our estimate of our minimum eight hits per pass, the
probability of getting the same return as this specific target over 6 consecutive pings during the one hour mission can be computed at approximately $3.7 \times 10^{-12}$. To put this number in perspective: if the BioSonar system was inspecting the totality of the surface of Earth (oceans and lands) we would have on average 1.5 false alarm for the totality of the globe. This shows that it is the target echo consistency provided by the BioSonar coupled with its wide beam pattern that holds the key to reducing false alarms.

3. TRIALS

3.1. Environment

During the trials we collected collocated sidescan images and BioSonar data from various sites in the inner and outer harbour area of Weymouth bay. In this paper we are particularly interested in a set of identical objects in the inner harbour area. This particular area has two different bottom types, a highly reflective sediment in the north part and a highly cluttered sediment to the south. Figure 3 displays two sidescan snapshots of the two different sediment types.

![Sidescan snapshot of the two different sediment types. (left) north, (right) south. Note the strong surface return in relatively shallow water to the south](image)

3.2. Targets

The main focus in this paper is a set of nine gas cylinder targets. These targets were laid in a harbour area. The cylinders are all identical in construction, see Figure 4 for an example. They measure 620 mm high by 300 mm diameter. Each end has a metal collar (bottom diameter 300 mm; top diameter 240 mm) and the cylinders were closed with square- head threaded stoppers. These structures provide additional scattering points and ensure that the targets do not exhibit a simple symmetrical cylinder response.

Prior to deployment three of these gas cylinders were water filled, three were filled with sand and the remaining three were filled with gravel with a 10 mm nominal particle size (indicative range 5-20 mm). The cylinders were laid horizontally on the seabed along a north-south line with approximately 50 m separating adjacent targets.
4. THE ATR PROBLEM

4.1. Target signature

Because of the geometrical configuration, broadside and end-on echoes from cylinder-like targets are much brighter than off-axis responses. Figure 6(a) displays a typical gas cylinder broadside response. The specular echo followed by secondary echoes are above reverberation level and clearly visible. Figure 6(b) shows an example of an off-axis gas cylinder response. The primary echo is only 1 or 2 dB above the noise level. Secondary echoes are often hidden within the noise.

4.2. Training data

Multi-aspect data were gathered in situ for the nine reference gas cylinders using spiral reacquisition profiles, see Figure 7, which notes the target positions and navigation transponder locations relative to the mission start point. Each three-turn track starts at nominal 25 m range.
and runs in to 20 m range. These profiles give a large number of target responses from all angles, though vehicle operating constraints limit the navigation accuracy and consequently impact on the number of usable returns.

**Fig. 6**: *Cylinder response*: (a) broadside response, (b) off-axis response.

**Fig. 7**: *Data gathering mission for multi-aspect responses of reference gas cylinders.*

### 4.3. SNR problem

The usual metric to assess signal quality is the SNR (Signal to Noise Ratio) value. For our application, noise includes system noise and reverberation level. The interpretation of the signal level strongly depends on the application. For detection purposes we fall into the classic interpretation of the SNR and the signal level represents the amplitude of the primary echo. For recognition application the target information lies within its secondary echoes and the SNR should then be interpreted as secondary echo level to reverberation level ratio.

Two main sets of features were used for detection and classification:

- identification method using a time-domain Gaussian mixture model to represent the matched filter echo [2].
- spectral features based on one-dimensional variant of Haralick’s co-occurrence texture feature set [3]. All calculations are based on normalised spectra, so intensity information is deprecated and classification is deterministically frequency dependent. Features are calculated across the band from 30-130 kHz.
4.4. Target recognition results

4.4.1 Detection

The gas cylinder targets are relatively small and give a poor response in the MarineSonics data, especially on challenging sediments. To illustrate the discrimination capability of the BioSonar, we compare with a set of objects selected to be of similar dimensions and echo strength to the gas cylinders. These objects are not readily distinguishable in the sidescan data. Figure 8(a) & 8(b) show the ROC curves respectively for single and two views of known cylinders and other targets with similar dimensions. Note that these curves cover views at any aspect.

![Image](image_url)

Fig. 8: ROC curve for (a) single view and (b) two views per target.

4.4.2 Identification

This is the next level in the identification chain and returns more information on the contents of the structure under investigation. In this case, we assume a first pass has identified an object as one of the gas cylinder targets and we now wish to extract information on the cylinder contents. In general HF has proved more effective in these trials. This seems to be more related to favourable SNR at these frequencies for the current system, rather than due to the physics underlying the classification process. Matrices for the fill discrimination are given in Table 1.

<table>
<thead>
<tr>
<th></th>
<th>Water</th>
<th>Sand</th>
<th>Gravel</th>
</tr>
</thead>
<tbody>
<tr>
<td>Water</td>
<td>93.39</td>
<td>0.00</td>
<td>6.61</td>
</tr>
<tr>
<td>Sand</td>
<td>8.10</td>
<td>49.63</td>
<td>42.26</td>
</tr>
<tr>
<td>Gravel</td>
<td>19.01</td>
<td>32.55</td>
<td>48.44</td>
</tr>
</tbody>
</table>

Table 1: Confusion matrices for classifier performance averaged over all aspect angles using HF transducers.

These figures indicate very good performance over all aspect angles (better than 90% correct in HF) in discriminating between the water-filled cylinders and sand/gravel fills.

4.4.3 Single-pass classification

Classification of single-pass returns from other missions backs up the figures given in the confusion matrix above, though results for discrimination of sand and gravel fills are generally better
than expectation based on these data. We would postulate that this is because the straight-line trajectory past an object typically gives more consecutive good SNR pings on target than are available from certain aspects within the training data given existing operational capabilities. This demonstrates the results we predicted in section 2.2.

Fig. 9: Classification distributions for single-pass sequences, HF channel: (a) 7 pings target T07 (water); (b) 10 pings on target T08 (sand); (c) 20 pings on target T09 (gravel).

Figures 9(a)-(c) show probability distributions for the first three contacts of the water, sand and gravel filled cylinders respectively. The water-filled cylinder echoes are very low level, but integrating 7 pings is sufficient to give distributions heavily weighted towards the correct class. The sand filled cylinder was at relatively short range (c. 10 m) and gives a completely unequivocal 'correct' response within 10 pings. The classification of the final cylinder is a tussle between sand and gravel, finally settling on gravel-fill after 20 pings, as illustrated in Figure 9(c)

5. CONCLUSION

In this paper we demonstrated the various capabilities of the BioSonar for MCM tasks in realistic scenarios. We have demonstrated the capability of such a system to learn in situ on difficult and low profile targets. We have demonstrated that contents of metallic cylinders can be distinguished at range from the wideband responses. We have demonstrated potential for single-pass classification and identification of low target responses and have shown that performance is greatly improved through the integration of target views from orthogonal passes - these can be achieved with a simple slalom mission profile through a target field.

REFERENCES


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Underwater Acoustic Measurement Facilities and Standards

Organizers: Anthony Paolero and Stephen Robinson
PROVISION OF STANDARDS AT SIMULATED OCEAN CONDITIONS

Graham A Beamiss, Stephen P Robinson, Gary Hayman

Abstract: The Acoustic Pressure Vessel (APV) has been established at NPL since 2000 and is used for characterizing underwater acoustic devices, determining the acoustic properties of viscoelastic materials and conducting integrity testing of artefacts while at simulated ocean conditions. The APV enables acoustic measurements to be undertaken at simulated ocean conditions: over a temperature range of 2 °C to 35 °C, and at hydrostatic pressures of up to 6.9 MPa.

This paper describes the APV facility and its operating capability, along with how underwater acoustic standards are provided. This includes how primary standard calibrations are realized, and how these are disseminated via comparison calibrations. The determination of the coefficients for transducer sensitivity variation as a function of temperature and pressure are an important factor in the dissemination process, both for measurements made in the APV and at the NPL open water facility (where temperature varies with the seasons). The methods adopted are described along with the influence of a number of factors which contribute to the uncertainty.
1. INTRODUCTION

1.1. UNDERWATER ACOUSTIC CALIBRATION

When products, such as acoustic systems containing acoustic transducers are sold, performance specifications quoted to the customer will often include absolute sensitivity (source level in the water is often of prime importance in determining range or detection limits), and the directivity (important for beam forming). Accurate measurement of these performance parameters is important for ensuring unambiguous specification and acceptance testing. Determining the acoustic performance requires acoustic measurements to be made [1], and for the measurements to be meaningful and they must be traceable to common standards of measurement [2, 3]. In the UK these traceable measurements are provided by the National Physical Laboratory (NPL), which is the only National Metrology Institute in Europe to be actively involved in underwater acoustics and able to provide such traceable measurements [4, 5]. The NPL acoustic pressure vessel (APV) addresses the requirement for free-field acoustic testing of transducers at simulated ocean conditions. The acoustic pressure vessel has been in full operation at the NPL following commissioning in 2000 [6-8]. It provides a unique facility in the UK, being the only facility of this type to be commercially available. The vessel forms an essential part of the UK’s National Measurement System for underwater acoustics.

1.2. TECHNICAL DESCRIPTION OF THE APV

The pressure vessel shown schematically in Figure 1 consists of a 7.6 m long by 2.5 m diameter tank and is manufactured from several sections of firebox steel. The test chamber simulates ocean conditions by having the capability to change water temperature and hydrostatic pressure in the range 2 °C to 35 °C and up to 7.0 MPa (equivalent to approximately 700 m of water depth) respectively. The test chamber weighs approximately 100 tonnes when it contains the 30,000 kg of water required to fill the chamber. Also highlighted in Figure 1 is the lining of acoustic absorbers that cover strategic sections of the interior vessel walls [9]. The absorbers appear to be quite suitable for this type of application, as their acoustic performance does not vary significantly over the operational specification of the test chamber.

There are two circular ports situated on top of the test chamber, which allow access for devices, materials or any artefact under test. The ports have access diameters of 0.50 m and 0.85 m respectively; the large port lid is capable of supporting devices weighing up to 500 kg and is equipped with a rotator shaft that passes through its centre. This allows for any artefacts attached to the mounting flange situated at the bottom of the shaft to be rotated through a full 360° while the vessel is at full pressure and also to be raised or lowered through a vertical height of approximately 130 mm, giving greater control and accuracy over acoustic alignment. Mounting processes involving heavy artefacts are carried out with the large port lid removed from the vessel port and seated securely on the lid rack system adjacent to the port. The small port lid does not have the ability to mount devices directly to it, but allows access to position pre-mounted devices to the internal slideway system. The slideway system has the capability to hold two lighter devices via mounting cradles and the entire slideway system may be traversed along the length of the
test chamber enabling positioning of devices to separation distances in the approximate range 1.0 m to 2.5 m

![Figure 1: Schematic diagram of the APV showing three-transducer spherical-wave reciprocity set-up.](image)

It is possible to achieve environmental conditions lower than sea state zero within the tank when employing the air mounts. A versatile operator control and monitoring system incorporating all non-acoustic tasks is run through a PC-based control system, providing live status of every piece of plant (water levels, flow rates, pressures and temperatures along with valve and pump conditions).

2. METHODOLOGY FOR TRACEABILITY

Calibrations of underwater acoustic transducers at simulated ocean conditions could be undertaken using an absolute method such as the method of three-transducer spherical-wave reciprocity [10]. This method has the advantage that it can be implemented with high accuracy and does not require a reference transducer that has already been calibrated. However, this method is time consuming to perform and requires at least three separate measurement arrangements. In the APV, this presents particular logistical difficulties since a lid must be removed before a transducer can be replaced. Although some accuracy is sacrificed, an alternative approach is to rely on relative calibration methods such as a comparison or substitution method. This allows the calibration to be performed much more rapidly, enabling a more efficient use of time in the APV. One disadvantage of relative calibration methods is that the reference transducers are required to have been previously calibrated over the full range of environmental conditions (temperatures and pressures) that are to be encountered in calibrations of unknown devices. This requires a great deal of initial work to characterise the reference transducers before calibrations may be attempted. In addition, it is not very satisfactory if the reference transducers used vary greatly in performance with temperature and hydrostatic pressure since large corrections will then have to be applied to the reference sensitivities, and this may introduce large uncertainties into the calibration process. Therefore, the ideal reference transducers are required to have relatively stable responses with temperature and pressure. Following evaluation of a selection of commercially available transducers and hydrophones, suitable devices were selected to be used as NPL reference devices that cover the operational frequency range of the test chamber (2 kHz to 400 kHz). The approach taken to
establishing and maintaining these devices as reference devices is described later in this paper.

3. ABSOLUTE CALIBRATION BY THE RECIPROCITY METHOD

The measurement methodology for free-field reciprocity is already established and is currently employed in the NPL open tank facilities [10-12]. However, significant alterations to many aspects of the existing procedure were required to enable this absolute method to be more suitably used within the APV.

The reciprocity method for absolute calibration requires the use of three transducers, designated as P, T and H, of which at least one must be reciprocal. The complete calibration requires the devices to be used in pairs during the three measurement stages, the paired measurement sequences being: P⇒T, T⇒P, P⇒H and finally T⇒H. For the APV, the procedure consists of elements taken from the two existing approaches currently in use in both of the open tank facilities to enable calibrations to be completed in a timely manner. A key element is the use of a dual hydrophone mount. With this, it is possible to have all three transducers in the test chamber at the same time, and still have the capability to rotate two of the devices out of the way when they are not being used in the measurement pair. A typical measurement set-up within the vessel is shown below in Figure 1, with transducer P being mounted through the small port and set-up so the transducers alignment mark faces in the direction of the two devices held beneath the large lid. Transducers T and H are mounted coaxially from mounting poles that are attached to the lid rotator via the dual hydrophone mount and they are initially set-up so that their respective alignment marks face towards the transmitting transducer P when they are rotated to the measurement position; with the measurement position being shown for devices P and T in Figure 1. It is possible to rotate transducers T and H through ±180° while the vessel is at full pressure and this ability enables three of the four required measurement runs to take place without any alterations being required to the three devices used in the calibration. Therefore, only one re-alignment of transducers is required to complete the calibration and that is to re-orientate transducers T and H about the vertical axis of their respective mounting poles so their respective alignment marks now face toward each other. The measurement process to fulfil a complete calibration is now as follows:

- Conduct measurement runs P⇒T and T⇒P
- Rotate large lid so that H is in the measurement position
- Conduct measurement run P⇒H
- Lift lid and re-align transducers T and H
- Conduct measurement run T⇒H

4. RELATIVE CALIBRATION METHODS

A reference transducer that has been calibrated and characterised as a function of temperature and pressure by an absolute method may be used to calibrate an unknown device using a relative calibration method. Such methods are between 2 and 5 times faster and better suited to performing multiple measurements at various temperatures and pressures in a time-efficient manner. Three relative methods are commonly used.
In the calibrated projector method, a reference projector is used to calibrate an unknown hydrophone [10]. The transmitting current response (TCR) of the projector has previously been determined by an absolute calibration by three-transducer spherical-wave reciprocity. By monitoring the drive current and knowing the separation between the projector and hydrophone, the sound pressure level at the hydrophone can be determined. A measurement of the electrical signal produced by the hydrophone will then enable the calculation of the receive sensitivity as follows:

\[ M_H = \frac{V_H D}{I_P S_p} \]  \hspace{1cm} (4.1)

where \( M_H \) is the hydrophone receive sensitivity, \( V_H \) is the receive voltage from the hydrophone under test, \( I_P \) is the projector drive current, \( D \) is the projector-hydrophone separation, and \( S_p \) is the transmitting current response of calibrated projector.

In the calibrated reference hydrophone method, a reference hydrophone is used to calibrate a projector under test [10]. This is effectively the reverse of the calibrated projector method described above. The measurement setup is the same except that this time the receive sensitivity of the reference hydrophone has previously been determined by an absolute calibration by three-transducer spherical-wave reciprocity. The sound pressure level produced by the projector under test can be calculated from the signal received by the reference hydrophone, the drive current and the transducer separation as follows:

\[ S_p = \frac{V_H D}{I_P M_H} \]  \hspace{1cm} (4.2)

If \( Z_p \) is the projector impedance, a measurement of the impedance of the projector will also enable its transmitting voltage response (TVR) to be calculated from:

\[ TVR_p = \frac{S_p}{Z_p} \]  \hspace{1cm} (4.3)

In the comparison method, a reference hydrophone is used to calibrate a hydrophone under test by a substitution method [10]. In the APV, this method may be used in combination with the dual hydrophone mount shown in Fig 1. so that three devices are mounted in the APV at the same time. A projector, which may be an uncalibrated device, is mounted via the small port. The reference hydrophone and the hydrophone under test are mounted through the large lid on each arm of the dual hydrophone mount. The hydrophones are mounted such that their reference directions are pointing away from the centre of the large lid. Initially, a measurement is made with the reference hydrophone facing the projector. The dual hydrophone mount is then rotated so that the hydrophone under test is in exactly the same position as the reference hydrophone was previously. A second measurement is made, and from these two measurements the receive sensitivity of the unknown hydrophone can be calculated.

\[ M_H = \frac{V_H}{V_{REF}} M_{REF} \]  \hspace{1cm} (4.4)

where \( V_H \) is the receive voltage from hydrophone under test, \( V_{REF} \) is the receive voltage from reference hydrophone, \( M_{REF} \) is the reference hydrophone sensitivity. If the projector is also a calibrated device, the measurement for the hydrophone under test may be performed as a calibrated projector measurement described above, thereby giving two
independent values for the receive sensitivity of the hydrophone. This provides a useful check of the method.

5. REFERENCE DEVICE COEFFICIENTS

Having selected suitable transducers and hydrophones, the following process is followed to establish and maintain them as reference devices for the facility:

- Absolute calibrations of the reference transducers using the primary standard method of free-field reciprocity is first undertaken to provide the reference sensitivity of the transducers at ambient pressure and 18 °C.
- Absolute calibrations of the reference transducers using the primary standard method of free-field reciprocity are undertaken at a series of pressures and temperatures that cover the required operational environmental conditions.
- The calibration results gained from this suite of calibrations is then used to calculate the pressure and temperature coefficients for each device.

By this approach, the reference sensitivities for the transducers are traceable to primary standards in the UK. Of course, it is not desirable for a reference devices performance to vary greatly with temperature and depth, as large corrections will then need to be applied to the reference sensitivities and this may introduce large uncertainties into the process. The plots shown above in Figure 2 show an example of the reference transducer coefficient data for selected APV reference devices. These devices have been chosen as demonstrate good stability when they are subjected to variations in the environmental conditions across the operational specification of the APV facility.

![Figure 2: Temperature coefficients for ITC 1001(left) and ITC 1089D(right).](image)

6. SUMMARY

Standards for underwater acoustics at simulated ocean conditions are provided in the UK by NPL using the Acoustic Pressure Vessel. The APV enables acoustic measurements to be made at hydrostatic pressures of up to just under 7 MPa (equivalent to about 700 m of water depth) and at water temperatures ranging from 2 °C to 35 °C. It is possible to calibrate and characterise underwater acoustic devices using both absolute and relative measurements methods. Relative methods are performed much more rapidly, enabling a more efficient use of time in the APV, but reference transducers previously calibrated over
the environmental conditions of interest are required to perform these measurements and some accuracy is sacrificed when compared with absolute methods.

7. ACKNOWLEDGEMENTS

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REFERENCES


RECENT ADVANCES IN METHODS FOR THE CALIBRATION OF LINEAR HYDROPHONE ARRAYS AT LOW FREQUENCY

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Abstract: Comparison calibrations of hydrophone data channels in linear arrays are challenged by a variety of factors. First, calibration of an array may require the simultaneous measurement of complex sensitivity (e.g., magnitude and phase) for data channels numbering in the hundreds and distributed over an extended aperture length. As a consequence of high channel counts and long electrical transmission paths, the acoustic data are frequently digitized within the array and transmitted over a telemetry system that cannot easily be synchronized with calibrated reference standard hydrophone data collected by commercial data acquisition systems. In addition, calibrations at low frequency are hampered by an inability to use pulsed waveform techniques to approximate free field propagation due to the arrival of boundary reflections before sufficient reflection-free data can be acquired. Finally, the minimum calibration frequency may be dictated by the operating bandwidth of the acoustic projector used to transmit the calibration waveforms. The Naval Undersea Warfare Center has developed methods to measure complex sensitivity in the data channels of hydrophone line arrays with digital telemetry systems. These methods have improved the quality of low frequency calibration data, while maintaining efficiency commensurate with a production test environment. A requirement for calibration at arbitrarily low frequency is that the equivalent electronic noise levels in the data channels are six to ten (or more) decibels below sea state zero; a requirement that is frequently satisfied in arrays designed for scientific, geophysical and naval applications.

Keywords: comparison calibration, hydrophone line array, low frequency
1. INTRODUCTION

The Underwater Sound Reference Division (USRD) is the organization sanctioned by the National Institute of Standards and Technology (NIST) to provide underwater acoustic calibration services that are traceable to United States and international standards. In this role, the USRD provides transducer and hydrophone calibrations for government, academic and industrial institutions throughout the full range of temperature and hydrostatic pressure found in the world’s oceans. In addition, USRD performs research and development to advance the measurement science in underwater acoustic metrology, including development of measurement methods for specific applications and facilities. Among these efforts is a project to develop methods for the measurement of complex sensitivity in the hydrophone data channels of line arrays at low frequency.

Secondary calibration of a hydrophone is often performed using a free field comparison method. [1] In this approach, a calibrated reference standard hydrophone with known free field voltage sensitivity is used to measure the acoustic field that is transmitted by an acoustic projector and compared to the output voltage of the unknown hydrophone. The free field voltage sensitivity of a hydrophone channel may be defined as the ratio of the complex voltage output by the hydrophone to the complex acoustic pressure in the immediate vicinity of the hydrophone. Thus, the free field voltage sensitivity is a complex quantity with both a magnitude and a phase, both of which may be required to realize the specified performance of advanced signal processing algorithms.

A fundamental requirement of traditional free field comparison calibration methods is the use of an acoustic projector to transmit a signal with an amplitude that is sufficient to overcome all sources of measurement noise including the ambient noise in the test environment, and the electronic self-noise of both the unknown and calibrated reference standard hydrophones. An additional requirement is that adequate quantities of acoustic data are collected prior to arrival of the first boundary reflection. These requirements limit the minimum frequency at which accurate measurements of complex sensitivity can be performed. The primary objective of this effort is to develop methods to measure the complex sensitivity of the hydrophone data channels in a line array when these requirements are not satisfied. A second objective is to perform the calibration measurements when the hydrophone array incorporates a digital acoustic telemetry system that cannot be accurately synchronized with acoustic data observed by the calibrated reference standard hydrophones.

2. MEASUREMENT METHODS

2.1. Traditional, Free Field Calibration

Among the challenges involved in the calibration of a hydrophone line array is the large size of the array itself. Hydrophone line arrays as employed in naval, scientific and geophysical applications have aperture lengths measured in the hundreds of meters. As a result, acoustic calibration of these arrays under well controlled laboratory conditions is not generally practiced, nor is it feasible. The USRD facility in Bugg Spring (Leesburg,
Florida) is an ideal site for the calibration of hydrophone line arrays due to favourable year-round weather conditions and near constant sound speed throughout a relatively large body of water. Figure 1a shows the general arrangement of the test site at Bugg Spring where a barge is located in a water depth of about 50 meters. The spring provides a slow continuous supply of fresh water that maintains a constant temperature and prevents the accumulation of algae or other biological fouling. The ambient noise level is typically equivalent to sea state zero or one, depending on local weather conditions.

A method for handling and calibrating hydrophone line arrays has been in use for several decades [2] where the array is affixed to a cylindrical test fixture and submerged into a body of water for calibration. An acoustic projector located on the central axis of the cylindrical test fixture is used to transmit calibration waveforms into the water where they are received by the data channels of the array and also by a calibrated reference standard hydrophone affixed to the test fixture. The test barge includes an overhead crane and array handling equipment to support the measurements. Data collection, processing and control equipment is housed in a climate controlled laboratory on the test barge.

Figure 1b provides a sketch of an array and test fixture as used during calibration measurements. A typical calibration is performed using gated continuous acoustic waveforms and well established comparison methods [1] for the calculation of channel sensitivity. Due to the characteristics of low frequency acoustic projectors and the geometry of the test setup, the low frequency limit for quality reflection free data is on the order of 100 Hz. At lower frequencies, the presence of boundary reflections, including those from the surface and test barge, cannot be avoided. While this test arrangement is known to introduce measurement uncertainties that are not encountered in a well-controlled laboratory environment, it enables the simultaneous calibration of hydrophone data channels numbering in the hundreds, resulting in considerable savings in both time and cost to calibrate an array. Alternative methods of calibrating individual array channels without use of the cylindrical test fixture are employed for special projects requiring greater precision, but can take several days to complete an array with an extended length aperture and high channel count.

Fig.1: NUWC-USRD Leesburg, Florida test facility, a) Bugg Spring geometry and, b) sketch of hydrophone line array assembled on test fixture (reproduced from [2])
2.2. Very Low Frequency Passive Calibration

A method to measure the complex sensitivity of the array data channels at frequencies less than the minimum effective frequency of available acoustic projectors was developed. The method uses the ambient noise in the natural environment as the source of the calibration signal. This approach is feasible when the equivalent pressure spectrum level of electronic noise in the array data channels and in the calibrated reference standard hydrophones are both substantially less than the ambient noise field, such as that illustrated in Fig. 2a. While this somewhat limits the field of applicability for this measurement method to low noise acoustic sensor systems, it is common for arrays designed for scientific, geophysical and naval applications to also be designed with low electronic noise levels. It is also standard practice to design calibrated reference standard hydrophones with electronic noise levels that are substantially lower than the equivalent ambient noise spectrum levels found in the intended operating environments.

An important requirement for complex calibrations based on the comparison of signals observed by an unknown array data channel and by a calibrated reference standard hydrophone is that the two signals are correlated. This requirement is satisfied in traditional calibration methods by using a projector to generate an acoustic field that is highly correlated throughout the volume occupied by the test equipment. In the absence of an actively transmitted calibration signal, observations of the naturally occurring ambient noise field must likewise be correlated.

The ambient acoustic field is assumed to result from an ensemble of identical noise sources uniformly distributed throughout the volume, and that the pressure produced by each source is a random, ergodic process with the same statistical properties. Analysis of the well-known, volume-noise model [3] for an isotropic ambient noise field shows that the correlation coefficient \( \rho_{12} \) for the acoustic field pressure observed at two locations separated by a distance \( d \) is

\[
\rho_{12}(kd) = \frac{\sin(kd)}{kd}
\]

(1)

where \( k \) is the acoustic wavenumber. Thus, for separations that are small relative to an acoustic wavelength (e.g., \( kd \ll 1 \)) the observed signals are highly correlated, with the correlation coefficient tending toward unity for diminishing frequency and separation (e.g., \( \lim_{kd \to 0} \rho_{12} = 1 \)). While this model does not account for the directional nature of noise generated by surface wind and waves, these sources contribute negligible power to the ambient noise spectrum observed at sufficiently low frequency. [4]

The greatest distance between any two points within the test volume was about three meters, a distance roughly corresponding to the diameter of the cylindrical test fixture illustrated in Figure 1b. Consideration of Eq. (1) shows that the theoretical cross-correlation coefficient \( \rho_{12} \) for ambient isotropic noise observed at two points that are separated by three meters in water with a sound speed of 1480 meters per second is greater than 0.9 for frequencies less than 60 Hz. Thus, the upper limit of the measurement band for the passive calibration was set to 60 Hz to ensure that the observed signals were
sufficiently correlated. The lower limit of the calibration band was 5 Hz as determined by
the minimum frequency at which calibration data were available for the reference standard
hydrophones.

The complex sensitivity $M_2$ of each hydrophone data channel in an array was
computed as the ensemble average of the acoustic transfer functions observed between
multiple calibrated reference standard hydrophones and the unknown array hydrophone as

$$M_2(f) = \frac{1}{N} \sum_{n=1}^{N} \frac{V_{12}(f)M_1(f)}{V_{11}(f)} e^{-j2\pi\tau}$$

(2)

where $V_{12}$ is the voltage cross-spectrum observed between one reference hydrophone and
one unknown array data channel, $V_{11}$ is the auto-spectrum observed by the reference
hydrophone, $M_1$ is the complex voltage sensitivity of the reference hydrophone, $N$ is the
number of observations, $f$ is frequency and $\tau$ is an interval of time representing the
relative delay between the different data collection and telemetry systems used for the
array (digital) data and the reference hydrophone data.

The relative delay $\tau$ between two low pass filtered, asynchronous signals was
determined using the cross correlation function of the reference $v_1$ and array channel $v_2$
voltages as

$$R_{12}(\tau) = \lim_{T \to \infty} \frac{1}{T} \int_{-T/2}^{T/2} v_1(t) v_2(t-\tau) dt$$

(3)

where $t$ is time. The filter cut off frequency was 60 Hz, corresponding to the upper limit
of the calibration bandwidth. Filtering was necessary to prevent corruption of the cross
 correlation calculation by directional, higher frequency components in the acoustic field.

Fig. 2: Acoustic field data observed during passive calibration of a hydrophone line array,
a) ambient noise spectrum in the calibration band and, b) cross-correlation coefficient
between an array data channel and a calibrated reference standard hydrophone.
Eq. (3) was normalized to yield the correlation coefficient $\rho_{12}$ as

$$\rho_{12}(\tau) = \frac{R_{12}(\tau)}{\sigma_1 \sigma_2}$$

(4)

where $\sigma_1$ and $\sigma_2$ were the standard deviations of the reference $v_1$ and array channel $v_2$ voltage signals, respectively. The relative delay $\tau$ between the two signals was found where the magnitude of the cross-correlation coefficient $|\rho_{12}|$ was a maximum as illustrated in Fig. 2b. In the event that the greatest magnitude was associated with a negative value, the polarity of the two voltage signals were known to be reversed, thus providing a method to quickly detect wiring errors in the array data channels.

### 2.3. Low Frequency Active Calibration

In principle, a comparison calibration can be performed with active acoustic signals transmitted by an acoustic projector despite the presence of boundary reflections in the calibration data. If the contribution from boundary reflections can be accounted for in a model of the total acoustic field at all points on the array, then that information can be included in calculations of complex sensitivity. [5] To test the feasibility of this approach, the test fixture was populated with eight calibrated reference standard hydrophones to sample the acoustic field over the cylindrical surface occupied by the array aperture.

The measurement was performed above the minimum effective frequency of 20 Hz for a Navy J15 projector by transmitting a continuous acoustic field with stationary statistics. The calibration signal was continuous, broadband Gaussian noise. Data collected by the reference hydrophones were processed using standard signal processing techniques to estimate the acoustic transfer functions between one of the reference hydrophones (e.g., principle reference) and the remaining seven reference hydrophones (e.g., auxiliary references) distributed over the cylindrical surface of the test fixture.

The acoustic transfer function $H_{pn}$ between the principle reference $p$ and the $n^{th}$ auxiliary reference located at azimuth angle $\theta_n$ and vertical displacement $z_n$, was estimated as the ratio of the cross spectrum $P_{np}$ between the reference hydrophones and the autospectrum $P_{pp}$ of the principle reference hydrophone using

$$H_{pn}(f,\theta_n, z_n) = \frac{P_{np}(f)}{P_{pp}(f)}.$$  

(5)

The frequency dependent, complex acoustic transfer functions were interpolated over the domain of azimuth $\theta$ and vertical displacement $z$ occupied by the reference hydrophones on a frequency-by-frequency basis. The result was an empirical, frequency dependent model of acoustic field variations on the surface of the test fixture, including the domain occupied by the array being calibrated, during transmission of a continuous, broadband calibration signal using a Navy J15 acoustic projector.
Figure 3 illustrates an example of the complex acoustic transfer function between the principle reference hydrophone and all other points on the cylindrical surface of the test fixture. The radiating surface of the acoustic projector was located on the longitudinal axis of the test fixture and at a vertical displacement of zero meters. The locations of the eight reference hydrophones are illustrated with cross-hatched circles.

Inspection of Figure 3a shows that the acoustic field magnitude lacked the axial symmetry that would be expected if the reflected acoustic field was due only to a pressure release surface located above the fixture. On the contrary, azimuthal variations of more than two decibels were observed at a vertical displacement of one meter. The corresponding phase variation was more than 15 degrees. The lack of axial symmetry in the complex acoustic transfer function was due to the influence of a prominent corner reflector formed at the intersection of the water’s surface and vertical surfaces of the nearby test barge. Therefore, by using an empirical model to account for spatial variations in the acoustic field, the measurement accuracy could be improved.

![Figure 3: Complex acoustic transfer function a) magnitude and b) phase estimated over the cylindrical surface occupied by the aperture of a hydrophone line array](image)

The complex, frequency dependent sensitivity of the $i^{th}$ sensor $M_i(f)$ in the array was calculated as

$$M_i(f) = \frac{V_i(f) \cdot M_p(f) \cdot e^{-j2\pi \tau}}{V_{pi}(f) \cdot H_{pi}(f, \theta_i, z_i)}, \quad (6)$$

where $M_p$ was the complex sensitivity of the principle reference hydrophone. The quantity $V_{pi}$ is the voltage cross spectrum between the principle reference hydrophone and the $i^{th}$ hydrophone in the array. The voltage autospectrum observed by the $i^{th}$ hydrophone of the line array was $V_{ii}$. The complex exponential accounts for the relative delay time $\tau$ between the data collection and telemetry systems for the array and the reference hydrophone data determined by analysis of the ambient noise field just prior to the active waveform transmission as was discussed in the previous section. The acoustic transfer function $H_{pi}$ between the principle reference standard hydrophone and the location of the $i^{th}$ line array hydrophone was estimated using the empirical model of the acoustic field variation on the surface of the test fixture as illustrated in Figure 3.
3. RESULT

The complex sensitivity measured in one data channel of an experimental hydrophone line array is provided in Fig. 4. The calibration result is provided as the average value in each of the standard 1/3rd octave bands spanning more than six octaves. Data used for the active calibration constituted about ten minutes of continuously transmitted broadband noise. Data used for the passive calibration was limited to a five minute ambient noise observation originally intended to characterize the test environment. Longer observation times should reduce the measurement variance for the passive calibration method.

![Complex sensitivity of an acoustic data channel in a digital hydrophone array, a) magnitude relative to design reference value and, b) phase relative to the incident acoustic field. Result represented as the ensemble average plus/minus one standard deviation.](image)

4. ACKNOWLEDGEMENTS

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REFERENCES

THE METHOD OF COMPLEX FREE-FIELD CALIBRATION OF A PRESSURE GRADIENT RECEIVER

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Abstract: Absolute free-field calibration of a pressure gradient receiver (PGR) is considered. The calibration is carried out using reciprocity method procedure during radiation of continuous linear frequency modulation signals in a water tank with reflecting boundaries. To obtain free-field frequency dependences the technique of complex moving weighted averaging is used.

The phase frequency response of pressure gradient receiver is obtained by including to the calibration procedure measurements phase angle of transfer impedance of the projector-receiver pair. To increase accuracy of phase frequency response measurements the position of the pressure gradient receiver acoustic center is defined by its displacement relative to pressure gradient receiver geometrical center.

Proposed method was used to calibrate at a frequency range 500 Hz – 12.5 kHz combined pressure gradient receiver which has three receive channels for vector quantity of acoustic field (pressure gradient) and one scalar channel (sound pressure).

The successful results of absolute complex free-field calibration of combined receiver allow to use phase closeness of instantaneous values of sound pressure and oscillation velocity in the same point of sound field as a plane wave criterion instead of traditional criterion based on 1/r law.

Analysis of results, advantages and new possibilities of proposed way for pressure gradient receiver absolute free-field calibration are discussed.

Keywords: pressure gradient receiver, calibration, the reciprocity method, phase angle of sensitivity, the complex moving weighted averaging.
Introduction

This article continues researching of abilities of technique of complex moving weighted averaging developed in VNIIFTRI [1]. Using CMWA technology solved the problem of overlapping frequency ranges of calibration in small volume camera and measuring water tank; allowed to extend frequency range up to 250 Hz of free-field measurement in reflected water tank with minimal size of 6 m [2, 3]. Attracting of new effective technique in a practice for free-field hydrophone calibration was the impetus for research on calibration of pressure gradient receiver (PGR) in reflected water tank by CMWA with chirp signal radiation.

Object for study

The object for study was a combined underwater sound receiver KGP 10 made by VNIIFTRI. It represents a sound pressure transducer and a three-component sound pressure gradient transducer mounted in one casing. A design of KGP 10 is shown in Fig. 1.

![Fig. 1: View of KGP 10 combined underwater sound pressure gradient receiver: (1) receiver, (2) dampers, and (3) ring.](image)

Three accelerometers made of a metal cube mass are mounted on the internal surface of the ball-like case of the receiver, which is 53 mm in diameter along three mutually normal X, Y, and Z axes; the cube is based on six double piezoelements. The switching circuit of the accelerometer provides three channels for receiving the pressure gradient and one channel for receiving the sound pressure.

Measurements description

The basis for the realization of absolute free-field PGR calibration become results of KGP 10 calibration by comparison method with a standard (reference) hydrophone. Measurements are performed in reflected water tank with dimensions 6×6×10 m by using CMWA method and complex chirp signals radiation. Before CMWA applying the measured frequency dependence were edited on a function, which represents the result of multiplying of the complex chirp projector current \( i(t) \) by the exponential frequency function of chirp signal \((\omega_0 + \mu t)^\alpha\) (\(\mu\) is the frequency variation rate) [4].
Fig. 2 shows the dependence of sensitivity of pressure gradient channel of PGR measured in water tank (1) and in a small volume chamber by the method of a vibrating column of liquid (2). These data are demonstrate good agreement between the sensitivities obtained by independent methods under different acoustic conditions. The straight line of best approximation of free-field frequency response (3) has a slope of 20 dB/decade, and deviations from the straight do not exceed 1.5 dB. The parallelism of frequency responses shows that the character of reception remains the same at high frequencies. Significant discrepancy between the results at frequencies above 630 Hz explained by sharp increasing pressure calibration error.

![Graph showing frequency dependence of sensitivity of PGR pressure gradient channel to sound pressure in field of equivalent plane waves measured in water tank (1) and obtained in a small volume chamber (2).](image)

For absolute PGR calibration a kind of procedure for complex reciprocity calibration method with three transducers in the traveling spherical wave was used. As a third transducer a calibrated PGR was used, which was treated as a directional receiver of sound pressure. Amplitude and phase responses of PGR channels to sound pressure gradient were calculated based on the link of sound pressure and sound pressure gradient in a spherical wave. Accuracy of measurements was controlled by stability of reversible transducer sensitivity previously measured at the hydrophone calibration.

**Calibration result**

Fig. 3 shows the frequency dependence of the sensitivity of channel $X$ of KGP 10 to the of the sound pressure gradient obtained by comparison method (1) and with the reciprocity method procedure (2).

Proximity of curves allows to speak as a good coincidence of the results obtained by independent methods, and high quality measurements in the field. Oscillation of frequency dependencies with a period of about 2 kHz indicate the source of scattering at a distance of about 74 cm from the acoustic center of the KGP 10 that corresponds to the position of the receiver sealed connector.

During $Y$ channel calibration, distortions of frequency response were detected, indicating phase non-identity due to accelerometers undercompensation. Fig. 4 shows the phase response of the channels $X$ and $Y$ of KGP 10, the behavior of which also demonstrates the existence of problems in the channel $Y$. 
Result analysis

During the free-field calibration some features can be revealed, which are impossible during the PGR calibration in a small chamber. For example, distortions of amplitude and phase frequency responses because of accelerometers undercompensation or due to influence of the PGR connector, which is always out of the chamber. Lack of opportunity to receive the phase response led to the fact that the vector receiver design was limited only by tuning of amplitude characteristics. Thus, the desired shape characteristics and identity of channels receiver achieved, in assumption that the necessary form and identity of phase characteristics were provided. Ability to obtain a phase characteristic allows the developer a tool for identifying of design features, defects and improving manufacturing technology at the stage of creating a prototype vector receiver.

Conclusions

The experimental results showed that the CMWA method with chirp signals radiation allows to allocate not only scalar, but also vector quantity of the direct wave field, which makes it possible to transfer the unit to vector receivers in conditions of their intended use – in the free field of a traveling sound wave. Successful absolute free-field calibration of a PGR
with the reciprocity method procedure has demonstrated the possibility of calibrating receivers of the vector parameters without a hydrophone with a known sensitivity. Ability to obtain a phase characteristic allows solving the problem of incomplete vector receiver pattern, attract phase response for the improvement of existing and development of new processing algorithm for amplitude and phase measurement.

REFERENCES

CALIBRATION OF HYDROPHONES IN THE FREQUENCY RANGE 1 KHZ TO 200 KHZ USING OPTICAL METHOD

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Abstract: A technique for absolute calibration of hydrophones using the optical method is described. In this method, the acoustic particle velocity is obtained at the air-water interface without a reflective pellicle using a commercial laser doppler vibrometer. The measurement principle is introduced and a calibration facility is set up. A B&K8105 hydrophone is calibrated over the frequency range from 1 kHz to 200 kHz using the calibration system. The result is compared with that of three-transducer reciprocity method and good agreement is shown. The measurement scheme eliminates the acousto-optic effect and pellicle mounting. The calibration is convenient to carry out.

Keywords: hydrophone calibration, optical method, laser vibrometry
1. INTRODUCTION

The sound pressure is one of the most important physical quantities in the underwater acoustic measurements. The measurement of sound pressure is usually carried on by hydrophones. The sensitivity of hydrophones must be calibrated in order to ensure the accuracy of pressure measurement. The reciprocal method [1] has been the preferred method for the calibration of hydrophones in the frequency range 1 kHz to 200 kHz. It is an absolute calibration method. According to the electro-acoustic reciprocity principal, this method needs a projector, a reciprocal transducer and a hydrophone. The sensitivity of the hydrophone can be determined with the knowledge of the electric parameters of transducers and acoustic field information. During the calibration procedure, three times of transducers mounting and alignment are needed, which may cause larger measurement uncertainty.

In the early 1980s, the optical method was used to calibrate the sensitivity of hydrophones. In 1988, National Physical Laboratory of UK investigated the calibration of hydrophones in the frequency range from 0.5 MHz to 15MHz using the optical method which is now the primary reference of UK [2].

The optical method is a non-invasive method to measure the acoustic parameters. When using the optical method to calibrate a hydrophone, a reflective and acoustic transparent pellicle usually need to be placed in the acoustic field [3][4]. If the thickness of the pellicle is less than the sound wavelength, the vibration of the pellicle is in phase with the particle velocity. The laser beam is incident to the pellicle to measure the displacement or velocity of the vibration. If the pellicle is in the far field of the projector, the sound pressure may be derived on the basis of the plane wave assumption. Then the hydrophone is palced on the same position as the pellicle measuring the open circuit voltage. The sensitivity of the hydrophone can be calculated according to the definition of hydrophone sensitivity. The existence of sound wave in water may change the local refractive index of medium, and this may change the optical path as the laser beam penetrates the acoustic field. The acousto-optic interaction may cause the measured displacement or velocity to be different from the true value. The acousto-optic interaction is dependent on the sound frequency and the acoustic field forms. In order to eliminate the effect of acousto-optical interaction and pellicle, a scheme for measure the particle velocity was investigated without the reflective pellicle. The laser beam propagates only in air and would not be modulated by the acousto-optic effect.

2. MEASUREMENT PRINCIPLE

2.1 Measurement arrangement

The measurement principal of particle velocity is shown in Fig.1. The projector is placed in water and the distance from the center point of the projector to the water surface is \(d\). The projector is excited with sinusoidal tone burst signals. The laser vibrometer head is suspended above the water surface. The laser beam is incident to the water and reflected on the water surface. The laser beam should be coincident with the acoustic axis of projector. The reflected optical signal contains the information of surface vibration velocity. The optical signal is detected by the LDV and the velocity signal is output by the decode circuit. The measurement velocity is related to the particle velocity off the \(d\) distance from the projector in the free acoustic field. The relationship between the water surface velocity \(v\) and particle velocity \(u\) will be derived. If \(d\) is larger than the far field distance, the free field sound pressure can be \(p = \rho cu\), where \(\rho\) is the density of water, and \(c\) is sound velocity in water.
To calibrate the sensitivity of hydrophone, the open circuit voltage $e_{oc}$ should be measured. The projector and hydrophone are mounted to the calibration frame. The projector and hydrophone are arranged to the proper position in the water tank to avoid the reflection from the surface.

### 2.2 Calculation of free field sound pressure

As is well known, the sound wave propagating at the interface of different medium would lead to reflection and transmission phenomenon [5]. As illustrated in Fig. 2, when the acoustic wave is incident to the boundary of medium with different characteristic impedance, the partial of energy is reflected and the other goes into another medium.

Let us consider the case when the sound wave is perpendicularly incident to the boundary. The arrow end denotes the propagation direction of waves. The wave equation in different medium can be expressed as:

\[
\begin{align*}
\frac{\partial^2 p_1}{\partial x^2} &= \frac{1}{c_1^2} \frac{\partial^2 p_1}{\partial t^2} (x \leq 0 \text{ half space}) \\
\frac{\partial^2 p_2}{\partial x^2} &= \frac{1}{c_2^2} \frac{\partial^2 p_2}{\partial t^2} (x \geq 0 \text{ half space})
\end{align*}
\]  

(1)
where \( p_1 \) is sound pressure in medium I, and \( p_2 \) is sound pressure in medium II.

The general solutions of the wave equation (1) are as follows:

\[
\begin{align*}
    p_1 &= (A_1 e^{-jk_1 x} + B_1 e^{jk_1 x})e^{i\omega t} \\
    p_2 &= (A_2 e^{-jk_2 x} + B_2 e^{jk_2 x})e^{i\omega t}
\end{align*}
\]

The particle velocity \( u \) and sound pressure \( p(x,t) \) are related as:

\[
u = -\frac{1}{\rho} \int \nabla p \cdot dt
\]

The particle velocity can be derived from equation (2) and (3),

\[
\begin{align*}
    u_1 &= \left( \frac{A_1}{\rho_1 c_1} - \frac{B_1}{\rho_1 c_1} \right) e^{-jk_1 x} + \left( \frac{A_1}{\rho_1 c_1} + \frac{B_1}{\rho_1 c_1} \right) e^{jk_1 x} \\
    u_2 &= \left( \frac{A_2}{\rho_2 c_2} - \frac{B_2}{\rho_2 c_2} \right) e^{-jk_2 x} + \left( \frac{A_2}{\rho_2 c_2} + \frac{B_2}{\rho_2 c_2} \right) e^{jk_2 x}
\end{align*}
\]

where, \( A_1 \) and \( B_1 \) are amplitude of forward wave and backward wave respectively in medium I; \( A_2 \) and \( B_2 \) are amplitude of forward wave and backward wave respectively in medium II.

There is no reflection wave in the half space where \( x > 0 \), thus \( B_2 \equiv 0 \).

\[
p_2(x,t) = A_2 e^{i(\omega t - k_2 x)} = p_1
\]

The pressure and normal particle velocity are consistent on the boundary (\( x = 0 \)), the following equation can be derived as:

\[
R = \frac{B_1}{A_1} = \frac{\rho_2 c_2 - \rho_1 c_1}{\rho_2 c_2 + \rho_1 c_1} = \frac{Z_2 - Z_1}{Z_2 + Z_1}
\]

\[
D = \frac{A_2}{A_1} = -\frac{2\rho_2 c_2}{\rho_2 c_2 + \rho_1 c_1} = \frac{2Z_2}{Z_2 + Z_1}
\]

\( R = B_1 / A_1 \) is defined as the reflective index, which is the ratio of reflective wave amplitude to the incident wave amplitude. \( D = A_2 / A_1 \) is defined as the refractivc index, which is the ratio of refractive wave amplitude to the incident wave amplitude.

When the sound wave travels from water to the air, \( \rho_2 c_2 \ll \rho_1 c_1 \), it means \( R \approx -1 \), \( D \approx 0 \), and \( A_1 = -B_1 \). The sound pressure \( p_1 \) and particle velocity \( u_1 \) can be expressed as:

\[
p_1(x,t) = A_1 (e^{-jk_1 x} - e^{jk_1 x})e^{i\omega t} = -j2A_1 \sin k_1 x e^{i\omega t}
\]

\[
u_1(x,t) = \frac{A_1}{\rho_1 c_1} (e^{-jk_1 x} + e^{jk_1 x})e^{i\omega t} = \frac{2A_1}{\rho_1 c_1} \cos k_1 x e^{i\omega t}
\]

The amplitude of reflective wave is equal to the incident wave, but the phase is reverse. The pressure on the boundary equals zero. The amplitude of particle velocity doubles that of
incident wave. The measured velocity \( v \) at the air-water interface by the laser vibrometer is two times of the particle velocity in the free-field at the same distance \( d \) off the projector. The free-field acoustic pressure can be expressed as:

\[
p = \frac{1}{2} \rho cv
\]  

(10)

The sensitivity of the hydrophone can be obtained by:

\[
m = 2e_\infty / (\rho cv) = 2e_\infty / (\rho cKU_v)
\]  

(11)

Where \( K \) is the sensitivity of LDV and \( U_v \) is the output voltage of LDV.

3. EXPERIMENTAL FACILITY AND RESULTS

The experimental calibration system includes transmitting subsystem, water tank, positioning subsystem, receiving subsystem and measurement software, etc. A B&K 8105 hydrophone is calibrated using the experimental facility. The laser vibrometer used is Polytec CLV1000. The auxiliary projector used are a spherical transducer with diameter 30 mm and a spherical transducer with diameter 60 mm, covering the frequency range from 1 kHz to 200 kHz. The measurement points are chosen at 1/3 octave frequency points.

The projector and hydrophone are mounted on a frame and placed in the water tank. The sinusoidal tone burst signals are used to excited the projector. The open circuit voltage signals are preamplified, filtered and the steady state signals are collected by a digital oscilloscope. Remove the hydrophone and adjust the distance from the projector to the water surface with the same projector hydrophone distance. Keep the transmitting conditions to be the same as before. The velocity signals from the LDV shares the same channel as the hydrophone signals. Then the sensitivity of the hydrophone can be calculated. The result using reciprocal method is shown in Fig. 3 for comparison.

\[\text{Fig. 3: Results comparison between optical method and reciprocal method}\]

As is shown in Fig. 3, the measurement results using the optical method are in good agreement with that of the reciprocal method in the frequency range from 1 kHz to 200 kHz. The maximal discrepancy is about 1.0dB at the frequency 4 kHz.
4. CONCLUSION

The calibration of hydrophones using the optical method is convinced in the frequency range 1 kHz to 200 kHz. The free field pressure is obtained by measuring the velocity at the water-air surface using a laser vibrometer without a pellicle. Compared with the reciprocal method, the optical method is more convenient to perform and it eliminates the effect of non-reciprocity and non-linearity of the transducer. Compared with the optical method using a pellicle, the referred method eliminates the acousto-optic effect and flextensional vibration of the pellicle, because the optical beam travels in the air and is reflected by the water-air interface, and it has the advantage of 6 dB gain in SNR.

REFERENCES

THE DESIGN OF ACOUSTIC ABSORBERS FOR TEST TANK LININGS

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Abstract: To provide good free-field region for measurements or reduce reverberation in the test tank, it is essential to use acoustic absorbers as linings. Generally, the absorbers are made from butyl rubber. In this paper, a new type of absorber using polyurethane is presented. Firstly, the design principle of wedge absorber is introduced, then its performance is analyzed theoretically, and the testing results are given finally. It is found that the wedge absorbers are suitable to be used as the tank linings when the frequency is in the range from 10 kHz to 200 kHz.

Keywords: underwater acoustics, wedge absorber, absorption coefficient
1. Introduction

Wedge absorbers have been widely applied in airborne acoustics and proved to be an outstanding absorption structure. In underwater acoustic measurements, by paving wedge absorbers as linings on the tank walls and bottoms, the acoustic reflections from these boundaries can be reduced significantly, and therefore the good free-field regions can be provided, and the reverberation times of water tanks can be shortened obviously.

Wedge absorbers in water tanks are usually made of rubber materials, the complicated production process and high costs hinder its extensive applications. Thus, the polyurethane was modified for different kinds of absorbers [1-3]. In this paper, the methods of preparing absorbing materials and making wedges are presented, the reflection factor of wedges is measured and the results validate that the wedges made from polyurethane with metal and foamed fillers can absorb acoustic waves sufficiently in the frequency range from 10 kHz to 200 kHz.

2. The preparation of absorbing materials

Consisted of hydroxyl (soft segment) and polyisocyanate (hard segment), the polyurethane has been found to be an ideal material for manufacturing wedge absorbers, and its acoustic properties can be improved by adjusting the ratio of hard and soft segments. Moreover, by adding metal filler in the material, the effect of sound absorption will be enhanced through vibrations of filler particles excited by acoustic waves. Generally, the graphite powder is chosen as fillers. By adding 20% graphite, the characteristic impedance of materials can be increased from $1.1 \times 10^5$ Pa.s.m$^{-1}$ to $1.3 \times 10^5$ Pa.s.m$^{-1}$, which become closer to the value of water. As a result, less acoustic waves will be reflected from the surface of the materials and more energy will be absorbed during the propagation of acoustic waves inside the materials.

To achieve wide-band absorbing materials and improve the absorption coefficient of polyurethane, the porous materials, i.e. vermiculite, are added in the polyurethane as bubbly fillers. These fillers will convert volume compression of elastomer into shear deformation, which will increase the friction of the materials dramatically. It is found that by adding 12% vermiculite, the absorption coefficient will increase about 9% in low frequency range. As the attenuation of acoustic waves is very sensitive to bubbles, the content of vermiculite must be controlled accurately.

3. The design of wedge absorbers

Generally, the absorbers are designed in shape of wedges. To contribute good absorption coefficient, the height of wedges should be larger than a quart of the wavelength at the lowest
working frequency. The incoming acoustic waves will be reflected continually on the surface of wedge. In the process, the acoustic energy will get into wedges and be absorbed gradually.

![Figure 1: Schematic diagram of wedge absorber](image)

To analyze the acoustic property of wedge absorbers, the gradient-transition absorption layer is used to approximately calculate the reflection factors of absorbers. The sound velocity \( c \) and density \( \rho \) of the absorbers are the function of height \( z \) (\( 0 \leq z \leq L_1 \)). When the shear elasticity of the material is neglected, they can be expressed as \(^{[4-5]}\):

\[
\rho(z) = \rho_0 + (\rho_1 - \rho_0)n(z)
\]

\[
c(z) = \left[ \frac{1}{\rho_0 c_0} + \left( \frac{1}{\rho_1 c_1} - \frac{1}{\rho_0 c_0} \right)n(z) \right]^{\frac{1}{2}}
\]

where \( \rho_0 \) is the density of water, \( c_0 \) is the sound speed of water, \( \rho_1 \) and \( c_1 \) are the density and sound velocity of the absorbing material, respectively, and \( n(z) \) is the function related to the shape of wedge that defined as the ratio of the sectional area at height \( z \) and the bottom area of the wedge. For the wedge shown in figure 1, \( n(z) = z/L_1 \), so the acoustic reflection factor \( R \) and the absorption coefficient \( I \) of absorbers can be approximated as \(^{[6]}\):

\[
|R| = \exp \left\{ -k_0 L_1 \text{Im} \left[ \frac{A + 2B}{2B} \sqrt{1 + A + B} - \frac{A}{2B} + 2 \frac{k_0}{k_1} \frac{L - L_1}{L_1} \right] \right\}
\]

\[
+ \frac{A^2 - 4B}{4B^{3/2}} \text{Im} \left[ \frac{A + \sqrt{B}}{A + 2B + 2 \sqrt{B \sqrt{1 + A + B}}} \right]
\]

\( A = k_0 c_0 \rho_1, \quad B = \frac{\rho_1}{\rho_0} \rho_1 \rho_0 \)
\[ I = 1 - |R|^2 \]  \hspace{1cm} (4)

where \( k_0 = \omega / c_0 \), \( k_1 = \omega / c_1 \), \( A = \left( \frac{\rho_0 c_0^2}{\rho_1 c_1^2} - 1 \right) + \left( \frac{\rho_1 - \rho_0}{\rho_0} \right) \), and \( B = \left( \frac{\rho_0 c_0^2}{\rho_1 c_1^2} - 1 \right) \left( \frac{\rho_1 - \rho_0}{\rho_0} \right) \).

To simplify installations, the wedge absorbers are usually designed as square boards, and the wedge absorber will be divided into four sections. The ridges of triangular cones are designed in parallel on each section and orthogonal to the others in adjacent regions, so that the acoustic waves in different incident angles can be absorbed evenly.

To produce wedge absorbers in frequency range from 10 kHz to 200 kHz, the height of wedge should be larger than 37.5 mm, and therefore the wedge absorbers with the length, width and thickness as 250 mm, 250 mm and 50 mm are manufactured, and the length, width and height of triangular cones are chosen as 125 mm, 30 mm and 40 mm, respectively.

Figure 2: Layout of triangular cones on wedge absorber

4. Measurements of wedge absorbers

4.1 Measurement method

There are different methods to determine the various parameters of acoustic materials \([7-8]\). In the frequency range from 10 kHz to 200 kHz, the reflection factor of absorbers can be acquired simply by measuring the signal reflected from samples pasted on tank wall. As shown in figure 3, the transducer is fixed into the water tank, and both the hydrophone and absorbers are located in the far field of the projector. To reduce the influences of reflected signals from other boundaries, the separation between the hydrophone and absorbers should
be set small enough. Tone burst signals are adopted so that both the reflected waves and direct waves can be recorded and analyzed simultaneously through digital scope after picked up by the hydrophone.

![Diagram](image)

*Figure 3: Arrangement for measuring reflection factor of wedge absorber*

The reflection factor $R$ can be obtained with the equation:

$$R = \frac{p_2}{p_1} \left(1 + \frac{2d_2}{d_1} \right)$$  \hspace{1cm} (5)

where $p_1$ and $p_2$ are direct-path and reflected signal received by hydrophone, $d_1$ is separation between projector and hydrophone, $d_2$ is separation between hydrophone and wedges.

### 4.2 Measurement results

The absorption coefficient of wedge absorbers made from polyurethane is measured and shown in figure 4. It reveals that the absorption coefficient can reach 85% at 10 kHz, and it gradually increases to 95% at 20 kHz. Though there are some fluctuations, the absorption coefficient is over 90% in the range from 20 kHz to 200 kHz.
5. Conclusions

The Polyurethane can be developed as sound absorbing material through modifying the components of materials, and adjusting the ratio of its hard and soft segments. By adding metal and bubbly fillers, the polyurethane can be used as absorbing materials in wide frequency range.

As a gradient-transition absorption layer, wedges can achieve good absorbing properties in related low frequency range. When the height of a wedge absorber is larger than a quart of the wavelength of sound, most of incoming energy will be consumed in the process of multiple reflections.

Through the theoretical analysis, it can be concluded that the wedge absorbers made from modifying polyurethane can achieve related higher absorption coefficient, and measurement results confirm that the wedge absorbers can be used as tank linings in the range from 10 kHz to 200 kHz.

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CALIBRATION METHODS OF THE INTERFEROMETRIC FIBER-OPTIC HYDROPHONE

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Abstract: The calibration methods of the phase-shifted sensitivity of the interferometric fiber-optic hydrophones are described and investigated experimentally. At first we introduced the measurement theories and their calibration facilities of three methods briefly: Bessel Function Ratio (BFR) method, Fringe Counting (FC) method, and Phased Generator Carrier (PGC) method. Then by the use of comparison calibration method, the calibration experiments of the phase-shifted sensitivity of a fiber-optic hydrophone based on the Michelson interferometer are carried on in a vibrating column vessel and free field respectively. By analyzing the experimental results, it is demonstrated that the phase-shifted sensitivity of the interferometric fiber-optic hydrophone can be calibrated correctly, by considering the different practical frequency range of these methods.

Keywords: Calibration method, fiber-optic hydrophone, phase-shifted sensitivity
1. INTRODUCTION

The fiber-optic hydrophone is different from the conventional piezoelectric hydrophone, which use the optical fiber as the sensing element. It is firstly proposed to detect the acoustic waves by Bucaro, et al [1] in 1977. Compared to the piezoelectric hydrophone, the fiber-optic hydrophone has its outstanding features of high sensitivity, immunity to electromagnetic interference and easy to scale into a large array, and has been greatly developed in the world wide [2]. A number of different optical hydrophone schemes have been developed based on the amplitude, phase, wavelength and polarization sensors. However, interferometric technology based on measurement of the optical phase changes which are proportional to the sound pressures has been proved to be both the most sensitive and the most practical. There are many kinds of optical fiber interferometers, but only the Michelson and Mach-Zehnder interferometers are push to practical application at the present.

In order to evaluate the performance of the interferometric fiber-optic hydrophones, calibration methods of Bessel Function Ratio (BFR) method, Fringe Counting (FC) method, and Phased Generator Carrier (PGC) method are studied by Takahashi [3], Xue [4], and Chen [5], et al. This article briefly introduces the principle of the fiber-optic hydrophone, measurement theories of three methods and their calibration facilities. Calibration experiments of a Michelson fiber-optic interferometric hydrophone are carried on in a vibrating column vessel and free field by using comparison calibration method. The experimental results demonstrate that phase-shifted sensitivity of the interferometric fiber-optic hydrophone can be calibrated correctly, but owing to the restriction of the techniques, the practical calibration frequencies of these methods are different.

2. PRINCIPLE OF FIBER-OPTIC HYDROPHONE

A schematic diagram of a Michelson fiber optic interferometric hydrophone is shown in Fig.1. A laser launches coherent light into an optical fiber coupler and then is split into two arms of the interferometer. One arm contains the sensing fiber coil and the other arm contains the reference fiber coil. The light of each arms is reflected by a Farad rotating mirror (FRM) respectively, and are recombined at the coupler, the output light is detected by an optical receiver and has the following form

\[ I = I_1 + I_2 + 2\sqrt{I_1I_2} \cos(\phi(t) + \phi_0(t)) \]  

(1)
Where $I_1$ and $I_2$ are optical intensities of two interfering arms respectively, $\phi_0(t)$ is the shifted optical phase generated by the sound pressure, $\phi_0(t)$ is the sum of initial phase and phase variation affected by the environmental interference, $t$ is the time. It is then be demodulated or displayed on the oscilloscope.

By measuring the $\phi(t)$ of an interferometric fiber-optic hydrophone and the sound pressure $p$ at the reference centre of the fiber-optic hydrophone in the undisturbed free field, the phase-shifted sensitivity can be calculated by:

$$M_\phi = \left| \frac{\phi(t)}{p} \right|$$ \hspace{1cm} (2)

The phase-shifted sensitivity level $M_\phi$ can be calculated by:

$$M_\phi = 20 \log_{10} \frac{M_\phi}{M_{\phi_r}}$$ \hspace{1cm} (3)

Where $M_{\phi_r}$ is the reference value, usually equals to 1 V/$\mu$Pa.

3. CALIBRATION METHOD

3.1 Measurement theories of shifted optical phase

3.1.1 Fringe counting method

In this method, the shifted optical phase $\phi(t)$ is got by directly counting the fringes of interference signal. If $\phi(t)$ can be expressed into $D\sin(\omega t)$, for convenience, equation (1) can be abbreviated as:

$$I = A + B \cos[D\sin(\omega t) + \phi_0(t)]$$ \hspace{1cm} (4)

Where $A$ is a DC term related to the loss of the optical intensity, polarization, etc. $B=kA$ is related to the optical intensity of interferometer, splitting ratio of coupler, reflectance and insertion loss of the FRM, etc. $k<1$ is the visibility of the interferometer.

For the optical interference signal is cyclical and regular, if $D$ equals to $n\pi$ ($n=1,2,\cdots$), the wave of interference signal looks like there is a cross line presenting on the wave along the time axis, and the number of stripes in half cycle equals to $n$. Thus, the shifted optical phase can be counted from the number of the optical interference fringes.

3.1.2 Bessel function ratio method

This method expands the optical interference signal into a fundamental term and many harmonic terms by using Bessel function, and the shifted optical phase $\phi(t)$ can be obtained by measuring the fundamental and harmonic terms and calculating their ratio. Therefore equation (4) can be expressed into:
Where $J_0$, $J_{2n}$, $J_{2n-1}$ are zero, $2n$ and $(2n-1)$ order Bessel function of the first kind respectively.

If the fundamental and third harmonic terms are selected and measured, the Bessel function ratio $\alpha$ can be calculated:

$$\alpha = 20 \log \left| \frac{J_3(D)}{J_1(D)} \right|$$

Thus, the amplitude of shifted optical phase $D$ can be obtained by using the lookup table or calculation. For a certain value of $\alpha$, there exists multi values of $D$. This can be solved through practical measurement or by the prior knowledge.

### 3.1.3 Phased Generator Carrier method

Basic theory of PGC method is to modulate the optical interference signal with a high frequency carrier, and get the shifted optical phase through demodulation.

After modulation, equation (4) can be changed into:

$$I = A + B \cos [C \cos \omega_c t + \phi(t)]$$

Where $C$ is amplitude of the carrier signal, $\omega_c$ is the angular frequency of carrier signal, $\phi(t)=D \sin(\omega_t t) + \phi_0(t)$ is the signal that need to be measured.

Differential cross multiplication and arctangent calculation are mostly used in demodulation. Figure.2 shows the flow chart of the PGC method base on arctangent calculation. Compared to the demodulation technique based on differential cross multiplication calculation, arctangent calculation has advantages of high sensitivity, wide dynamic range, and its measurement result is not susceptible to the low frequency drift.

![Figure 2 Flow chart of PGC method base on arctangent calculation](image)

### 3.2 Measurement theories of sound pressure [6]

In this article, comparison calibration method with a standard hydrophone is used for measuring sound pressure. Measurement of sound pressure in the frequency range 20 Hz to 1 kHz is carried on in a cylindrical stainless steel vessel by using vibrating column comparison method.

During measurement, the standard hydrophone and the fiber-optic hydrophone are simultaneously immersed in the column of water with their reference centers at the same horizontal plane in a cylindrical vessel, which is excited externally by an electro-dynamic transducer at the bottom. The sound pressure which applied on the fiber-optic hydrophone can be calculated by:
\[ p = \frac{U_0}{M_0} \]  

Where \( U_0 \) is the open-circuit voltage of the standard hydrophone, \( M_0 \) is the calibrated sensitivity of the standard hydrophone.

Measuring sound pressure in the frequency range 1.25 kHz to 10 kHz is carried on in an anechoic water tank of 50 m long, 15 m wide, and 10 m deep. The free field comparison calibration method is used. The standard hydrophone and fiber-optic hydrophone are simultaneously placed in the far field of a projector. The sound pressure which applied on the fiber-optic hydrophone can be calculated by:

\[ p = \frac{U_1d_1}{M_0d_2} \]  

Where \( U_1 \) is the open-circuit voltage of the standard hydrophone, \( d_1 \) and \( d_2 \) are the distance between the reference centers of the projector and the standard hydrophone, and the distance between the reference centers of the projector and the fiber-optic hydrophone respectively.

4. CALIBRATION FACILITIES

The schematic diagram of calibration facility using FC method in the free field with a standard hydrophone is shown in Figure.3. The calibration facility is consist of an acoustic signal transmitting system, an acoustic signal receiving system and an optical signal transmitting & receiving system.

![Figure.3 Schematic diagram of calibration facility using Fringe Counting method](image)

When BFR method is used for calibration, the measurement instruments of the acoustic signal transmitting system and optical signal transmitting & receiving system are kept in same, the oscilloscope of acoustic receiving system is changed to a dynamic signal analyzer for analyzing the harmonic interference signal.

When PGC method is used for calibration, the measurement instruments of the acoustic signal transmitting system and acoustic signal receiving system are kept in same, an optical phase modulator and a phase demodulator are added to modulating and demodulating the interference signal.

5. CALIBRATION RESULTS AND CONCLUSION

The calibration results of a Michelson fiber optic interferometric hydrophone are shown in Figure.4. Where \( M_1, M_2, M_3 \) are respectively the phase-shifted
sensitivities calibrated by using PGC method, FC method, and BFR method. Owing to the restriction of the calibration techniques, the calibration frequency ranges of PGC method, FC method, and BFR method are 20 Hz ~ 10 kHz, 31.5 Hz ~ 10 kHz, and 125 Hz ~ 1 kHz. The calibration results of these three methods are agreed very well, the maximum deviation is only 0.9 dB at 10 kHz.

Figure 4 Calibration results of a Michelson fiber optic hydrophone

From the experimental results, it can be concluded:
(1) The phase-shifted sensitivity of the interferometric fiber-optic hydrophone can be calibrated correctly by using PGC method, BFR method, and FC method.
(2) FC method is the most direct and convenient technique with its clear physical concept. Its calibration frequency range is wide. But it is easily affected by the external vibration.
(3) The measurement using BFR method is relative complicate. Due to the multi values in calculation, prior knowledge of the applied signal is needed. This method is also easily affected by the external vibration. Because the dynamic signal analyzer used is not fit for measuring the pulse signal, calibration in free field is not carried on.
(4) The calibration technique using PGC method is the most complicate, it has the high accuracy of measurement results through eliminating the external interference by modulation and demodulation processing. But it is difficult to extend to more high frequency range due to the limitation of the carrier frequency.

REFERENCES

THE CALIBRATION AND CHARACTERISATION OF AUTONOMOUS UNDERWATER RECORDERS

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Abstract: Methodologies for the calibration and characterisation of autonomous recorders used for in-situ measurement of underwater noise are presented. The increasing use of autonomous recorders is motivated by the need to monitor underwater noise, such as in response to the requirements of the EU Marine Strategy Framework Directive, and the aim is to provide traceability for underwater noise monitoring to underpin the protection of the marine environment from anthropogenic noise. The performance of these systems is a crucial factor governing the quality of the measured data. In this paper a discussion is presented of measurement methodologies for key acoustic performance characteristics of the recorders, including the self-noise of the hydrophone and system and the absolute sensitivity as a function of frequency (including hydrophones, amplifiers and digitisation system). Consideration is given to effects due to the proximity of the recorder body to the measuring hydrophone: where the hydrophone is attached close to the recorder body, scattering from the body can have a significant influence on the frequency and directional response of the overall system. Examples of the results obtained are given and a discussion is presented of the implications of system performance on the quality of the measured data.

Keywords: calibration, noise measurement
1. INTRODUCTION

Concern over the potential damage that can be caused to the marine environment by noise from anthropogenic activity has led to a greater demand for measurement of noise in the ocean and coastal waters. In order to satisfy this demand a range of autonomous underwater recorders has appeared on the market over recent years. Where these devices are used for tasks requiring no absolute measurement, such as monitoring the presence of marine mammals, knowledge of the sensitivity of the system may not be required. However, for measurements of absolute levels of ambient or man-made noise it is vital that the performance of the overall system, in terms of sensitivity, directional response, self-noise and dynamic range, is known. While the various methods for calibrating hydrophones are well documented [1, 2], there is currently no standardisation of the methods used to characterise autonomous recorders.

The UK National Physical Laboratory (NPL) is currently working on a government funded project to establish calibration techniques for autonomous acoustic recorders. This paper summarises the issues specific to the performance characterisation of such devices, describes methodologies investigated during the project and presents some of the results obtained.

2. MEASUREMENT METHODOLOGIES

The measurement chain of a typical autonomous recorder will generally consist of three stages: hydrophone, pre-amplifier and A/D converter. Consequently, it is necessary to determine the overall sensitivity of the complete system. In the calibration of hydrophones, the electrical signal produced by the device in response to the acoustic pressure can be measured directly. Unfortunately, this is not the case with an autonomous recorder, where the electrical signal is digitised and saved, typically in .wav file format, to on-board storage media and has to be analysed later. The output of such a system is in digital counts and there is an argument that the unit of sensitivity for such systems should be based on this (e.g. digital counts per pascal). However, to determine the sensitivity in units commonly used for hydrophones, i.e. V/Pa or dB re 1 V/µPa, the full-scale voltage range of the A/D converter needs to be known in order to translate the levels in the recorded data into actual voltages.

2.1. Pistonphone

The sensitivity of a hydrophone can be determined by comparison to a reference microphone using an air pistonphone [2]. The unknown hydrophone and microphone are inserted into an air-filled chamber and simultaneously exposed to the same acoustic pressure. It is a method that can only be used for low frequencies where the sensitivity of a hydrophone is the same in air as in water. In practice, the upper frequency limit is generally determined by the size of the chamber, in that its dimensions must be small enough in relation to the acoustic wavelength for the sound pressure to be regarded as constant throughout the chamber. This upper limit is typically around 300 – 400 Hz.

The pistonphone can be used to calibrate an autonomous recorder by mounting the recorder hydrophone in the coupler along with a calibrated reference microphone and
subjecting both devices to single tonal signals over the required frequency range. The recorder is set to record the entire frequency sweep and, as the signals are being generated, the microphone output at each frequency is measured. After the completion of the sweep, the recorded data are analysed and the voltage levels at each frequency determined by applying the relevant scaling factor. The sound pressure in the pistonphone coupler at each frequency can be calculated from the microphone output voltage and sensitivity. The overall sensitivity of the recorder can then be determined by taking the quotient of the recorded voltage and the calculated sound pressure.

2.2. Free-field measurements

NPL has two open-tank facilities in which free-field calibration of hydrophones can be performed. Free-field conditions are approximated by the use of tone-burst signals and windowing techniques to ensure that reflections from the medium boundaries do not affect the measurements. The larger of the two tanks is 5.5 m in diameter and 5 m deep, enabling free-field calibrations to be performed from high kilohertz frequencies down to around 1 kHz. In this facility the sensitivity of an autonomous recorder can also be measured through the use of a calibrated projector. By monitoring the drive voltage into the projector and knowing its transmitting voltage response and the device separation, the sensitivity of the autonomous recorder system can be determined from analysis of the waveform files recorded during the measurements. The mounting carriages in the facility also have rotational capability and so, by use of a combination of stepped-frequency and stepped-angular measurements, the frequency and directional response of the recorder can be characterised simultaneously.

2.3. Self-noise

The noise floor of an autonomous recorder, determined by the self-noise of the various components of the system, is an important performance characteristic which limits the lowest signal that can be measured by the recorder. This is particularly crucial if the recorder is to be used to measure very low-level ambient noise. Previous measurements have shown that, while a high quality hydrophone can have a noise floor close to the Wenz minimum level, the self-noise of an autonomous recorder can often be around, or above, sea-state zero level [3].

Measurement of the self-noise of an autonomous recorder and hydrophone requires the use of an acoustically-isolated room or chamber which provides no external acoustic stimulus. Autonomous recording systems are typically battery-operated and so isolated electrically from the mains supply but, if necessary, measures may need to be taken to isolate the system under test from any electrical pick-up from the surrounding environment. It is possible for systems to pick-up radiated electrical signals from the circuitry within the recorder, but this is considered to be part of the self-noise of the system being measured. The recorder is placed in the acoustically-isolated chamber and set to record for a period of a few minutes. At NPL, the recorded waveforms are analysed using in-house software written in the Matlab® programming language, which uses the system sensitivity to determine the power spectral density and/or third-octave band power levels over the measured frequency range.
3. RESULTS

The frequency and directional response of an autonomous recorder were characterised using the NPL large open-tank facility. The measurements were made in two configurations: one with the measuring hydrophone fixed directly to the end cap of the recorder; the other with the hydrophone separated from the recorder body by a 2.5 m cable. Stepped-frequency tone-burst signals from a calibrated projector were recorded by the autonomous recorder at each angle as it was rotated from -90 degrees to +90 degrees in 2 degree steps; the zero degree position being such that the measuring hydrophone was end-on to the transmitted signal. The recorder was mounted horizontally in such a way that the active element of the hydrophone remained on the axis of rotation at all angles.

![Comparison of frequency response of recorder with fixed and cabled hydrophones when measured side-on (-90 degrees).](image)

Fig. 1: Comparison of frequency response of recorder with fixed and cabled hydrophones when measured side-on (-90 degrees).

Fig. 1 shows the overall sensitivity of the recorder measured over the frequency range 1 kHz to 100 kHz. The response is plotted for each hydrophone configuration measured with the device side-on to the transmitted signal (-90 degree position). It can be seen that, in this orientation, the frequency response is generally similar for the two configurations. This is not the case for the end-on orientation (0 degree position), where Fig. 2 shows significant differences in the frequency response measured in the two configurations. While the response measured with the cabled hydrophone is relatively smooth and comparable in level to that seen from the side-on orientation, the response measured with the fixed hydrophone shows significant interference effects. The source of the interfering signal can be estimated by calculating the path difference between the direct signal and the interfering signal from the difference in frequency between the troughs (or peaks) in the
response [4]. In this case the path difference is consistent with the extra distance travelled by the acoustic wave from the hydrophone to the recorder end-plate and back again, suggesting that the interference is due to reflections from the recorder body.

Fig. 2: Comparison of frequency response of recorder with fixed and cabled hydrophones when measured end-on (0 degrees).

The effect of reflections from the recorder body in the fixed hydrophone configuration can also be seen in the directional response of the recorder. Fig. 3 shows the ± 90° directional response measured at 18 kHz for each hydrophone configuration. Both plots are normalised to the peak level measured for the fixed hydrophone configuration. It can be seen that, for this particular frequency over the range ± 25°, reflections from the recorder end-plate interfere constructively with the direct acoustic signal to produce a higher than expected response. This suggests that, if the recorder were deployed vertically with the hydrophone pointing upwards, acoustic signals at 18 kHz emanating from a source directly above the recorder would appear to be significantly higher than the true level if the side-on (or nominal) system sensitivity were to be used in the analysis of the measured data. If such a source were loud, producing a sound pressure level close to the upper measurable limit of the recorder, there could also be potential for saturation of the measured signal.
Fig. 3: Comparison of directional response of recorder with fixed and cabled hydrophones at 18 kHz.

Fig. 4 shows the ± 90° directional response measured at 44 kHz for each hydrophone configuration. Again, both plots are normalised to the peak level measured for the fixed hydrophone configuration. In contrast to the 18 kHz response, over a similar angular range, reflections from the recorder end-plate are now interfering destructively with the direct acoustic signal to produce a much lower than expected response. For the vertical deployment scenario described above, there would effectively be a “blind spot” directly above the recorder at 44 kHz, with the measured level appearing significantly lower than the true level if the data were analysed using the side-on (or nominal) system sensitivity. For a quiet source, sound produced close to 44 kHz could potentially be below the noise floor of the recorder and therefore not be seen in the measured data.
4. DISCUSSION

The design of many autonomous recorders is such that the measuring hydrophone is mounted directly, or very close via a short cable, to the recorder body, which is typically an air-filled cylinder. As can be seen from the results presented above, the proximity of the recorder body to the measuring hydrophone can have a significant influence on the frequency and directional response of the overall system. (One potential drawback of the pistonphone method is that only the hydrophone is exposed to the acoustic signal and so any effects from the recorder body are not taken into account. However, for the frequency range over which this method can be used, it is expected that the in-water wavelength will be large enough for the influence of the recorder body to be minimal. The low-frequency response of a recorder would therefore be expected to be similar to that of the cabled hydrophone i.e. close to omnidirectional.) The effect of perturbations in the response at kilohertz frequencies due to scattering from the recorder body can be minimised to some extent if the recorded noise data is analysed as third-octave band levels; in which case a degree of averaging of the frequency response over the band will occur. Similarly, for ambient noise measurement, where the recorder is measuring noise from all directions, peaks and troughs in the directional response will tend to be averaged out to some degree. The question is then what value of system sensitivity to use for the third-octave band analysis, as this will vary with both frequency and direction. Ideally, where the system is used for the measurement of ambient noise, the diffuse field response of the recorder may be required.

![Fig.4: Comparison of directional response of recorder with fixed and cabled hydrophones at 44 kHz.](image-url)
5. ACKNOWLEDGEMENTS

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REFERENCES


LONG TERM UNDERWATER THIRD OCTAVE SOUND LEVELS AT A BUSY UK PORT

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Abstract: The sound levels of the third octave bands with centre frequency 63 Hz and 125 Hz will be used as indicators for the EU Marine Strategy Framework Directive (MSFD) descriptor 11 to monitor low frequency continuous sound in the marine environment. To explore this, long term underwater sound data from a busy UK port were investigated. Two autonomous multichannel acoustic recorders (AMAR Generation 2; Jasco Applied Sciences) recording broadband sound in the effective frequency range 10 Hz to 32 or 48 kHz, for half an hour in every hour, have been deployed alternately at the Falmouth Bay Test site for marine renewable energy devices (FaBTest) off the south coast of Cornwall, UK from March 2012 to August 2013. Data collected during periods of wave energy device testing were removed for this analysis. The area supports considerable commercial shipping and recreational boating along with diverse marine fauna, including bottlenose dolphins, harbour porpoises and fish. Custom MATLAB scripts were used to derive third octave levels (TOLs). The mean half hourly TOLs were found to vary by season with a mean TOL for the 63 Hz band increasing from 87.88 dB re 1 µPa in July 2012 to a mean of 98.10 dB re 1 µPa for December 2012. The yearly mean TOLs of 92.39 ± 8.45 dB re 1 µPa and 95.14 ± 6.61 dB re 1 µPa for the 63 and 125 Hz bands respectively (number of half hour sound files = 6992) were below the suggested target of 100 dB re 1 µPa for the period March 2012-March 2013. This provides information on the current sound levels from which a trend can be monitored at this site. The empirical data presented here offers an exploration of the proposed MSFD indicator bands in order to inform future use.

Keywords: MSFD, third octave levels, shipping noise
1. INTRODUCTION

Low frequency vessel noise may affect marine species by increasing stress (Wysoczki, Dittami & Ladich 2006; Rolland et al. 2012) or masking communication signals and environmental cues (Clark et al. 2009). Masking has been found to occur in laboratory conditions (Kastelein et al. 2009) and several groups of animals have been documented to be affected by non-impulsive low frequency noise in the wild, including mysticetes (Parks, Clark & Tyack 2007; Anderwald et al. 2013), seals (Anderwald et al. 2013) and fish (Vasconcelos, Amorim & Ladich 2007). It is considered that the low frequency ocean sound levels are increasing, mainly due to increases in commercial shipping (Andrew et al. 2002).

The recognition of anthropogenic underwater sound as a potential driver of negative impacts for marine species has led to it being included in the Marine Strategy Framework Directive (MSFD) as a descriptor which should be monitored in order to determine if the seas of the member states have Good Environmental Status (GES) (The European Parliament and the Council of the European Union 2008). This is challenging to do for this descriptor as there is limited evidence on the current sound levels, trends and effects on marine species of any increased sound levels. Therefore, it has been recommended that further research is done to assess the current levels and trends in member states’ seas (Van der Graaf et al. 2012).

For this research, underwater sound levels have been recorded at a marine renewable energy device testing facility in Falmouth Bay, UK since March 2012. The dataset has been analysed for the third octave sound levels (TOLs) at the site.

2. METHOD

2.1 Location

A passive acoustic monitoring (PAM) device was deployed at the Falmouth Bay test site (FaBTest) site on Cornwall’s south coast, UK. The site is 2 km² in size, between 3 and 5 km offshore and 20 to 50 m in depth.

Falmouth Harbour is a busy commercial port with 1,193 ship arrivals reported in 2012 (Department for Transport 2013) which is the second highest in the South West. The area also supports considerable recreational boating (Latham et al. 2012). There is also diverse marine fauna including bottlenose dolphins, harbour porpoises, seals and fish (Pikesley et al. 2011).

2.2 Data gathering equipment

Two Autonomous Multichannel Acoustic Recorders (AMAR Generation 2; Jasco Applied Sciences) were deployed at FaBTest, alternately, on seven occasions, of which six are reported on here due to a loss of the data from the sixth deployment, at a distance of ~200 m from a wave energy converter. These devices use GeoSpectrum M8E hydrophones which were calibrated by Jasco Applied Sciences prior to deployment. The AMAR was programmed to record for the first 30 minutes in every hour. Table 1 shows the deployment locations and equipment settings. The effective frequency range is 10 Hz to half the sampling frequency. There were two methods of deployment; the dome configuration and flotation collar configuration. The resolution of the recording is 24-bit.

For the dome method, the AMAR was attached to a custom built steel frame. The frame was weighted and rested on the seabed. For the flotation collar method, a flotation collar was
attached around the AMAR which causes it to float in vertical position in the water column. The AMAR was attached to the centre of a weighted ground rope and was approximately 5 m off the seabed. During deployments three to seven the hydrophone cage was covered with a cloth hat to reduce flow noise.

<table>
<thead>
<tr>
<th>Deployment number</th>
<th>Date of Deployment</th>
<th>Position (\text{(Degrees and Decimal Minutes)})</th>
<th>Sampling frequency (\text{(kHz)})</th>
<th>Deployment method</th>
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<td>1</td>
<td>10(^{th}) March 2012</td>
<td>N50.099720 W04.996390</td>
<td>96</td>
<td>Dome</td>
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<tr>
<td>2</td>
<td>13(^{th}) June 2012</td>
<td>N50.098889 W04.995278</td>
<td>64</td>
<td>Flotation collar</td>
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<tr>
<td>3</td>
<td>20(^{th}) August 2012</td>
<td>N50.100409 W04.996118</td>
<td>64</td>
<td>Flotation collar</td>
</tr>
<tr>
<td>4</td>
<td>8(^{th}) November 2012</td>
<td>N50.100633 W04.995900</td>
<td>64</td>
<td>Flotation collar</td>
</tr>
<tr>
<td>5</td>
<td>09(^{th}) January 2013</td>
<td>N50.101256 W04.996308</td>
<td>64</td>
<td>Flotation collar</td>
</tr>
<tr>
<td>7</td>
<td>04(^{th}) June 2013</td>
<td>N50.100283 W04.997333</td>
<td>64</td>
<td>Flotation collar</td>
</tr>
</tbody>
</table>

Table 1: Deployment date, locations and equipment settings of the AMAR

As there were different deployment configurations which could have affected the received levels (RLs), for example, the hydrophones were at different depths for the dome deployments and flotation collar deployments, the data were kept separate for analysis. As the sound levels are recorded at a wave energy device testing site, the periods of testing are removed for this analysis.

### 2.3 Data processing

Following retrieval of the AMAR, the data were downloaded from the device to a computer and converted to .wav files using proprietary software.

The acoustic data were calibrated using the hydrophone response curves provided from the manufacturer’s calibration and interpolated to provide a calibration value per Hz, with an acoustic gain of 0 dB.

MATLAB scripts were developed to process the .wav files. These include a fast Fourier transform (FFT) function, using a 1 s Hann window with a 50% overlap, performed for every file to provide the power spectral density (PSD) in square pressure \(p_{\text{RMS}}^2\). The mean of the square pressure values were calculated per minute per Hz and stored.

To calculate the third octave levels, the square pressure values were summed together per minute, within the frequency range 57-71 Hz and 113-141 Hz for the third octave bands with 63 Hz and 125 Hz as the centre frequency respectively. The values are then converted to dB once all processing and averaging is completed. The mean square pressure, or arithmetic mean, has been used in line with the latest known recommendation (Van der Graaf et al. 2012).
3. RESULTS

3.1 Half hourly mean third octave levels

The sound levels range from 61.13 to 124.36 dB re 1 µPa and from 70.56 to 121.16 dB re 1 µPa for the 63 and 125 Hz bands respectively.

![Graph showing half hourly mean third octave levels](image)

**Fig. 1: Mean half hour third octave levels for the 63 Hz and 125 Hz bands from square pressure ($p_{RMS}^2$). 1st-7th indicates deployment number**

<table>
<thead>
<tr>
<th>Period</th>
<th>Mean sound level ($dB$ re 1 µPa ± STD)</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>63 Hz TOL</td>
</tr>
<tr>
<td>1st-7th deployments</td>
<td>91.90± 8.39</td>
</tr>
<tr>
<td>1st-5th deployments</td>
<td>92.39± 8.45</td>
</tr>
<tr>
<td>2nd-7th deployments</td>
<td>91.81± 8.23</td>
</tr>
<tr>
<td>3rd-7th deployments</td>
<td>92.64± 8.03</td>
</tr>
</tbody>
</table>

**Table 2: Mean sound levels for different periods of time**

Table 2 shows long term mean sound levels. Deployments 1 to 7 include all the data collected. Deployments 1-5 cover a year from March 2012-March 2013. Deployments 2-7 cover the deployments all utilising a flotation collar. Deployments 3-7 include all the deployments which used both a flotation collar and cloth hat to reduce flow noise.

3.2 Monthly mean third octave levels

![Graph showing monthly mean sound levels](image)

**Fig. 2: Mean monthly sound levels for the 63 Hz and 125 Hz bands.**
Fig. 2 shows the arithmetic mean for the monitored period by month. The number of half hour files used to calculate the average for March and August 2013 were 12 and 76 respectively. The number of half hour sound files used for the remaining months ranged from 380 to 741.

The gaps in the data (Fig. 1) are caused by either: removal of the data during periods of testing of the wave energy converter, lack of suitable weather windows which prevented access to the FaBTest site or the loss of the data from the 6th deployment.

4. DISCUSSION

There is greater variation in the 63 Hz third octave band than the 125 Hz band (Fig. 1; Table 2). The average sound levels are higher in the 125 Hz band than in the 63 Hz band (Table 2). Although the 125 Hz includes the summed energy for a greater bandwidth, in a previous study the sound levels at 125 Hz have been found to be louder than at 63 Hz with a bandwidth of 1 Hz (Garrett, Witt & Johanning 2013), supporting the suggestion that the sound levels are louder at 125 Hz at this site.

Most ships produce sound energy at frequencies less than 1000 Hz (Merchant et al. 2012; van der Schaar et al. 2014). Both distant shipping and intermittent local vessel traffic have previously been found to affect the sound levels in Falmouth Bay with energy from these sources concentrated in the frequency range 0.01-1 kHz (Merchant et al. 2012). However, the peak frequency of shipping noise was found to be 315 Hz (Merchant et al. 2012) which would not be included in these proposed indicator bands.

The third octave levels vary by month, increasing from the summer to the winter for the 125 Hz band (Fig. 2). The mean monthly sound levels varied by ~10 dB and ~7 dB for the 63 Hz and 125 Hz bands respectively over the whole period. Therefore, when measuring the yearly sound levels for the MSFD, samples should be taken throughout the year, covering all seasons.

The sound levels for the 3rd to 7th deployments (August 2012 to August 2013) had identical deployment configurations utilising a flotation collar and hat. For this period, the yearly mean sound level is 92.64 dB and 95.97 dB re 1 µPa for the 63 and 125 Hz bands respectively (Table 2). These levels are below the initially proposed target of 100 dB (Tasker et al. 2010). This provides information on the current sound levels from which a trend can then be monitored (van der Schaar et al. 2014) at this site.

ACKNOWLEDGEMENTS

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OBSERVATION OF AMBIENT NOISE INDUCED BY THE INTERNAL SOLITARY WAVE IN THE CENTER OF KUROSHIO NORTHEAST OF TAIWAN

Yiing-Jang Yang, Wen-Der Liang, Jeff Chih-Hao Wu, Hsian-Chih Chan, Ruey-Chang Wei, Chi-Fang Chen

Abstract: A subsurface acoustic Doppler current profiler (ADCP) mooring was deployed in summer of 2012 to monitor the ambient noise and current velocity on the I-Lan Ridge, northeast of Taiwan. The water depth at the mooring was 275 m. The mooring was close to the northeasterward Kuroshio mainstream. Several internal wave packets were recorded during spring tide periods. When the wave passed the mooring, the ADCP instrument dived into deep water and temperatures increased. A southwestward current accelerated with this downwelling as well as the temperature rise, and then followed by an opposite pattern in the upper ocean; therefore, a mode-1 depression internal solitary wave packet can be easily identified. The maximum horizontal current perturbation and vertical amplitude were around 150 cm s⁻¹ and 40 m, respectively. The mode-1 depression waves could generate the surface wave breaking in the convergence zone and produce ocean noises. The ocean ambient noises would be enhanced in the convergence zone and reduced in the divergence zone due to rough and smooth sea surface conditions, respectively. This study observed the sound source from surface wave breaking affected the high sonic bands between 100 Hz and 10 kHz. The maximum noise was around 10 dB increment at 1.5K Hz and coincided with the maximum convergence (downwelling), which agrees well with the theory. Strong horizontal currents induced by internal solitary waves would produce self-noise in the infra and low sonic bands below 300 Hz. Meanwhile, internal solitary waves can change the thermal structure and modify the aspects of the
underwater acoustic signal. Temperature profiles will be required to establish an underwater ambient noise model for further study.

**Keywords:** Kuroshio, internal solitary wave, underwater ambient noise
1. INTRODUCTION

Internal solitary waves are localized internal gravity waves occurring in a stratified ocean. These waves greatly impact the ocean environment as well as underwater acoustic propagation. Most observed internal solitary waves are categorized as mode-1 waves. A mode-1 internal solitary wave displaces isotherms downward in deep water, but upward when the upper layer becomes thicker than the lower layer. The former type is known as a depression wave, and the latter, an elevation wave [1]. The surface wave breaking generated by internal solitary waves in the convergence zone can produce ocean noises and its associated sea surface signatures can be imaged by radars [2], [3]. In addition, internal solitary waves would change the thermal structure as well as the sound-speed fluctuations. Therefore, the aspects of underwater acoustic propagation would be modified by the internal solitary waves [4]. Recent review papers by Helfrich and Melville [1] and Apel et al. [5] showed that internal solitary waves not only altered the underwater acoustic propagation, but also changed the ambient noises. These sea surface noises would be enhanced in the convergence zone and reduced in the divergence zone due to the rough and smooth sea surface conditions, respectively. In this paper, we will study the internal solitary wave in the center of western boundary current, Kuroshio, and its impact on underwater ambient noises. This paper is organized as follows. Section 2 is observation. A general description of the internal solitary wave theory and data analysis results are presented in Section 3. Finally, Section 4 provides discussion.

2. OBSERVATION

A subsurface mooring was deployed from May 7 to June 24, 2012 to monitor the current velocity and underwater ambient noises on the I-Lan Ridge, northeast of Taiwan (Fig. 1). The water depth at the mooring was 275 m. The mooring was close to the northeastward Kuroshio mainstream according to the composite current velocity at 30 m shown in Fig. 1 [6]. This mooring equipped an upward-looking 300-kHz broadband self-contained Acoustic Doppler Current Profiler (ADCP) at 100 m, a Several Hydrophone Recorder Units (SHRU), and two SBE39 Temperature-Pressure recorders (TPs) paired with the ADCP and SHRU, respectively. In order to avoid the cable strumming noise, we used the Kevlar rope with hair for the mooring lines at upper and lower of SHRU. The ADCP recorded current information from 26 m to 94 m in 4-m bins every 6 seconds and then averaged to 1 minute. The TPs sampled every 1 minute. Sampling rate of the SHRU was set to 19531.25 Hz, sensitivity of hydrophone was -170 dB re $\mu$Pa/V, and fixed linear gain was +26 dB. The power spectral density (PSD) of the SHRU data was calculated using a short-time Fourier transform with FFT and window points set to 1024. No overlapping was applied. An averaged PSD used in this study was then obtained by averaging all PSD in each record (about 1 minute).

Fig. 2 shows an example of an internal solitary wave packet, which passed the mooring site during 19:00-21:00 on June 6, 2012. This wave packet included 4 waves around 19:10, 19:50, 20:20, and 20:50, respectively. When the wave passed the mooring, the ADCP and SHRU instruments dived into deep water but temperatures increased. A southwestward current acceleration occurred with a downwelling as well as a temperature rise and then followed by an opposite pattern in the upper ocean; therefore, a mode-1 depression internal solitary wave packet can be easily identified. Acoustic data shown in
the bottom panel of Fig. 2 also revealed louder noises when the wave packet passed, especially during the first two leading waves. A 5-minute time lag existed between the noise peaks in the high and low sonic bands indicative of different physical processes.

![Fig. 1: The mooring location (shown as a red dot) on the I-Lan Ridge northeast of Taiwan. The local depth was 275 m. Bathymetry is color shaded and overlaid on composite current velocity vectors at 30 m from Liang et al. [6]. The reference vector is 50 cm s⁻¹.](image1)

**Fig. 1:** The mooring location (shown as a red dot) on the I-Lan Ridge northeast of Taiwan. The local depth was 275 m. Bathymetry is color shaded and overlaid on composite current velocity vectors at 30 m from Liang et al. [6]. The reference vector is 50 cm s⁻¹.

![Fig. 2: From top to bottom panels show the color contour plots of the east (U), north (V), and vertical (W) velocity components, temperature, depth and acoustic power spectral density, respectively, during 18:00-21:00, June 6, 2012. The black and red lines represent the TP data on the ADCP and SHRU, respectively.](image2)

**Fig. 2:** From top to bottom panels show the color contour plots of the east (U), north (V), and vertical (W) velocity components, temperature, depth and acoustic power spectral density, respectively, during 18:00-21:00, June 6, 2012. The black and red lines represent the TP data on the ADCP and SHRU, respectively.

Fig. 3 shows time-depth contours of horizontal and vertical current perturbations and time series of acoustic data during the first leading internal solitary wave on June 6, 2012. Horizontal and vertical current components were relative to 18:49 on June 6, 2012, before the passage of the internal wave. The horizontal current data was rotated to 212°T, the wave propagating direction. The maximum horizontal current perturbation occurred at 19:09 at 44 m while maximum perturbations of downwelling and upwelling were at 19:06, and 19:13, respectively, at 80 m. The maximum PSD in high sonic bands coincided with the downwelling, but the maximum PSD in low sonic bands were about 2-minute lagged behind the maximum horizontal current perturbation.

The evidences of internal solitary wave in the center of Kuroshio northeast of Taiwan not only from in-site mooring data but also from satellite images. Fig. 4 is a Moderate Resolution Imaging Spectroradiometer (MODIS) image captured at 05:05 GMT on 25 May 2013 (spring tide). Signals revealed two ISW packets with “dark-bright” bands around 24°35’N, 122°E. According to the curvature of the leading wave and subsequent waves, these two wave packets were primarily propagating SW and SWS, respectively.
3. INTERNAL SOLITARY WAVE THEORY AND DATA ANALYSIS

Based on weak nonlinear theory, the KdV equation, which includes a quadratic nonlinear term, is commonly used to describe the waveform of an internal solitary wave. The horizontal current velocity \( u \) analytical solution of K-dV equation involves a squared hyperbolic secant function, \( u(\xi, z) = U(z) \text{sech}^2(\xi) \), where \( \xi = x - ct/\Delta \), \( t, x \) and \( z \) denote the time, and horizontal and vertical directions, respectively, \( U(z) \) is amplitude, \( c \) is nonlinear phase speed, and \( \Delta \) is nonlinear characteristic width [7]. The vertical current velocity \( w \) can be calculated from the continuity equation. Therefore, \( w \) is proportional to the convergence/divergence value \( (\partial u/\partial x) \) at each layer. Furthermore, we can calculate \( \partial w/\partial x = 0 \) (or \( \partial^2 u/\partial x^2 \)) to find the points of the extremum. At these two extremum points, \( x = \xi_m \), the horizontal current velocity is \( u(\xi_m, z) = 2U(z)/3 \). It is inferred that the maximum downward or upward current speed occurring with the horizontal velocity value is two thirds of the horizontal current amplitude at each layer.

Since the surface wave breaking generated by internal solitary waves in the convergence zone produces ocean noises, the maximum ambient noise would occur with the maximum convergence value as well as the maximum downward current speed or two thirds of the horizontal current amplitude. In the divergence zone, the sea surface is smoother and more silent than the initial condition.
Fig. 5: Upper panel: The time series of the 1506 Hz (black) and 2499 Hz (blue) acoustic power spectral density and vertical velocity perturbation at 80 m (red) during 18:50-19:30 on June 6, 2012. Lower panel: The time series of the 19 Hz acoustic power spectral density (black) and the horizontal current speed perturbation at 48 m (red) during 18:50-19:30 on June 6, 2012.

Fig. 6: Top and bottom panels show the color contour plots of the along-wave, and vertical current perturbation components, respectively, during 18:50-19:30, June 6, 2012. From left to right columns overlaid the DJL solutions with small amplitude without background current, large amplitude without background, and large amplitude with uniform background current conditions, respectively.

Time series of current and acoustic measurements in Fig. 5 shows that maximum ambient noises in high sonic bands (e.g. 1506 and 2499 Hz) occurred simultaneously with the maximum downwelling at 80 m. The variation of PSD in high sonic bands was around 10 dB. This result agreed well with the theoretical analysis that the maximum underwater ambient noise would be related to the maximum convergence. The noises at infra and low sonic bands were pseudo-noise resulting from the presence of the hydrophone and its supporting structure in a current. This self-noise is also called the “flow-noise” [8]. The variations of acoustic in low sonic bands could be related to the current speed around the hydrophone. In this observation, we did not have current data at SHRU’s depth. Nevertheless, temporal variations of the horizontal current speed induced by mode-1 internal solitary waves were consistent in the upper and lower ocean but in opposite directions, so the current speed in the upper layer could represent the lower layer current speed. The variability of current speed at 48 m was similar to that of 19 Hz PSD as shown in the bottom panel in Fig. 5 except the acoustic data delayed for about two minutes.

Besides, the mode 1 internal solitary wave solutions of the full nonlinear equations include background currents, can be obtained solving the Dubreil-Jacotin-Long (DJL) equation [9], [10]. The background currents would be modified the wave width and phase speed [10]. Applied the DJL equation to examine the Kuroshio affected the internal
solitary wave. Fig 6a shows the solutions of DJL equation associate with small amplitude and without background current conditions. Compared with the observation data, the time scale of wave is the same order but current speed is too slow. Opposite of Fig 6a, Fig 6b shows the solutions of DJL equation associate with large amplitude and without background current conditions. The current speed is the same order with observation data but time scale is too small. The solutions of DJL equation associate with large amplitude and uniform background current (80 m s$^{-1}$) are show in the Fig 6c. The internal solitary wave direction is against the background current. Under this condition, the wave speed would be slow down and increased the time scale. These solutions are closed to the observation results and indicate that the Kruoshio affected the properties of internal solitary wave.

4. DISCUSSION

The time lag between the 48-m current speed and 19 Hz PSD (Fig. 5) could be related to the tilt of the mooring line. The current perturbation in the lower ocean induced by southwestward propagating mode-1 depression internal solitary waves were northeastward and strengthened by the northeastward Kuroshio. When these mode-1 depression internal solitary waves passed the mooring, the ADCP on the top of the mooring was pushed further northeast than the SHRU which was in the middle. The southwestward propagating internal solitary waves arrived earlier to the ADCP than SHRU. The horizontal distance between ADCP and SHRU was around 40 m and the phase speed of the wave was around 35 cm/s, so the propagation time was around two minutes. This agreed with the observed time lag. Comparing the 1506 Hz and 2499 Hz PSD with the 2-minute-set-back 19 Hz PSD, we found that the peaks at 1506 Hz and 2499 Hz PSD and two thirds of the peak at 19 Hz PSD occurred almost simultaneously. This result consisted with the previous theoretical analysis.

![Fig. 7: The acoustic power spectral density at 18:50 (black), 19:06 (red), 19:11 (blue), and 19:20 (green) on June 6, 2012.](image)

The whole bands of PSD before wave arrival, at the maximum of convergence, maximum of current speed, and after wave passed are illustrated in Fig. 7. Self-noises in the infra and low sonic bands below 300 Hz were affected by the flow while underwater ambient noises in high sonic bands between 100 Hz and 10K Hz were induced by sea surface wave breaking. The maximum noise was around 10 dB increment at 1.5k Hz.

Figs. 5 and 7 also show a temporal offset between the minimum noise and the maximum divergence (or upwelling) due to the contamination of the sound source from sea surface wave breaking. Nystuen [11] reported that 50% of sound energy arrives from an area equal to $\pi h^2$ over the hydrophone, where $h$ is the depth of the measurement. When the maximum of divergence passed the mooring, the horizontal distance between the maximum convergence and the SHRU was about 150 m, close to the SHRU depth.
Therefore, the most silence would delay for a while until the convergence zone far away from the SHRU. Meanwhile, internal solitary waves can change the thermal structure and modify the aspects of the underwater acoustic signal. Temperature profiles will be required to establish an underwater ambient noise model for further study.

5. ACKNOWLEDGEMENTS

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INFORMATION OF OCEAN BOTTOM BACKSCATTERING MATRIX PROVIDED BY SINGLE-MODE REVERBERATION


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Abstract: The characteristics of scattering due to interface roughness are usually described by the backscattering matrix (BSM). The BSM based on the Bass perturbation theory has significant differences from that based on the popular empirical scattering law (Lambert’s law), especially at low grazing angles. In an experiment with a point source, it is very difficult to extract the quantitative characteristics of the BSM at low grazing angles from the experimental data because of the difficulties in acquiring low-grazing-angle scattering data and separating the scattering data between different modes (grazing angles). In contrast, the use of single-mode excitations as sources in shallow-water waveguides enables acquisition of good quality low-grazing-angle scattering data. In this paper, the characteristics of the BSM were obtained from single-mode reverberation experiment in shallow-water. Model-data comparisons were made and the results showed that at low grazing angles (2°~5°), the BSMs based on the Bass perturbation theory were in good agreement with the experimental data, but the BSMs based on Lambert’s law were not.

Keywords: Bottom backscattering matrix; single mode; low grazing angles
1. GENERAL

Shallow-water reverberation modeling usually consists of three components: the propagation of the incident field in the waveguide (the Green’s function); calculation of the back-scattering matrix (BSM); and the propagation of the backscattered field. Since the 1980s, the Green’s function has been described by normal mode theory, but the critical component in the reverberation modeling, the BSM, has been treated by introducing an empirical scattering law \[1\] or using an assumed model with unknown parameters to be determined \[2\]. The popular empirical scattering law is Lambert’s law. The BSM, SR, based on Lambert’s Law is

\[
[S_{mnR}]^2 = \mu \sin \theta_m \sin \theta_n, \tag{1}
\]

where \(\mu\) is the so-called scattering coefficient, which is a constant and not related to the grazing angle, and \(\theta_m\) is the grazing angle corresponding to the \(m\)th mode. The proper BSM should be derived by resolving the integral equation expressed by the Green’s function. However, there is still no analytical solution to this equation. Recently some researchers derived the BSM under the Born approximation based on the Bass perturbation theory \[3\]

\[
[S_{mnR}]^2 = \mu \sin^2 \left(\frac{P}{2} \theta_m\right) \sin^2 \left(\frac{P}{2} \theta_n\right), \tag{2}
\]

where \(P\) is a bottom parameter related to the bottom reflection phase shift.

Obviously, at low grazing angles, there is a significant difference between the BSMs, and thus the reverberation levels given by Eqs. (1) and (2). In the traditional reverberation experiment with a point source, it is very difficult to obtain the BSM at low grazing angles from the reverberation data. Instead, an effective approach to acquire low-grazing-angle bottom scattering data corresponding to a given mode is to use a single-mode excitation and receiving system. In shallow-water waveguides, the single-mode field is usually excited by a linear vertical source array (VSA). In order to excite the single-mode field, each element of the VSA should be weighted properly. The vector of weights (VW) is dependent on the environmental parameters and the center frequency of the signal. Recently Peng et al. \[4\] discussed a closed-loop excitation method that can obtain the optimum estimation of the matrix of the Green’s function in the shortest feedback times, with high efficiency of emission of the single mode signal. The single-mode reverberation experiment in this paper is based on this optimum closed-loop excitation method.

This paper is organized as follows. The difference in the reverberation level between the first and second modes is analyzed based on the Bass perturbation theory in Section 2. Next the details about the single-mode reverberation experiments are introduced and the model-data comparison is made in Section 3. Finally, the summary and conclusions are addressed in Section 4.

2. THEORY ANALYSIS

According to the Bass perturbation theory, the scattered field of a pulse signal \(s(t)\) due to the bottom roughness is \[3\]

\[
u_i(R_0, R; t) = \frac{2ni}{k_0 r_c} s(t - t_c) \sum_m \sum_n \varphi_m(z_0) \varphi_n(z) \exp[-(\beta_m + \beta_n) r_c] S_{mnK}^R(k_m, k_n). \tag{3}
\]
and the average intensity of the reverberation contributed by the incoherent summation is
\[ I_R(R_0, R; t) = \left( \frac{2\pi}{k_0 c_0} \right)^2 s^2 (t - t_c) (2\pi r_c) \sum_{l=0}^{\infty} \sum_{m} \phi_m^2(c_0) \phi_n^2(z) \exp[-2(\beta_m + \beta_n) r_c] (S_{mn}^R)^2 \Gamma(k_m, k_n), \]  
(4)

where
\[ t_{mn} = \left[ \frac{\partial k_m}{\partial \omega} + \frac{\partial k_n}{\partial \omega} \right] r_c = \left[ \frac{1}{u_m} + \frac{1}{u_n} \right] r_c \approx \frac{2r_c}{c_0} = t_c, \]  
(5)

\[ S_{mn}^R = \phi_m(H)C_{mn}^R \phi_n(H) = C_{mn}^R \sin \left( \frac{P}{2} \theta_m \right) \sin \left( \frac{P}{2} \theta_n \right), \]  
(6)

\[ C_{mn}^R = \left[ k_0^2 - k_n^2 \right] / \alpha + (1 - \alpha^{-1}) k_m k_n + (1 - \alpha) a^{-2} \gamma_m \gamma_n, \]  
(7)

\[ \gamma_m = (k_m^2 - k_n^2)^{1/2}, \]  
(8)

\[ K_m(k_m, k_n) \equiv \int dr_1 \eta(r_1) \exp[i(k_m + k_n)r_1], \]  
(9)

\[ \Gamma(k_m, k_n) \equiv \frac{e^{c_0 \sigma_\eta^2}}{2} \int_0^\infty dx R^n(x) \exp[i2k_0x] = \frac{e^{c_0 \sigma_\eta^2}}{2} p^n(2k_0). \]  
(10)

The correlation function \( R^n(x) \) and the mean square (MS) value \( \sigma_\eta^2 \) characterize the interface roughness fluctuations, \( \eta \), respectively. \( P^n \) is the power spectrum of \( \eta \), \( \tau_0 \) is the duration of the signal \( s(t) \), \( H \) is the depth of the water, \( c_0 \) and \( c_b \) are the sound speeds in the water and the bottom respectively, and \( \rho_0 \) and \( \rho_b \) are the densities in the water and the bottom respectively. \( r_c \) is the center range of the scattering area (we assumed that the scattering area has limited size), and \( \beta_m \) is the modal attenuation.

If both the incident and scattered fields are the \( l \)th single-mode fields, Eq. (3) becomes
\[ u_{ll}(R_0, R, t) = \left( \frac{2\pi}{k_0 c_0} \right)^2 s(t - t_c) \cdot e^{-2\beta_1 r_c} \cdot S_{ll}^R \cdot K_m^R(k_1, k_l), \]  
(11)

and the average intensity of the reverberation is
\[ I_{ll}(R_0, R, t_c) = \left( \frac{2\pi}{k_0 c_0} \right)^2 s^2(t - t_c) (2\pi r_c) e^{-4\beta_1 r_c} [S_{ll}^R]^2 \Gamma(2k_0). \]  
(12)

According to Eq. (12), we obtain the difference of the reverberation level between the first and second modes
\[ 10 \log \left( \frac{I_{ll}}{I_{11}} \right)_{\text{pert}} = 10 \log \left( e^{-4(\beta_2 - \beta_1) r_c} \frac{S_{ll}^R}{[S_{11}^R]}. \right) \]  
(13)

For close ranges, the exponent factor in Eq. (13) is close to 1, and Eq. (13) can be simplified as
\[ 10 \log \left( \frac{I_{ll}}{I_{11}} \right)_{\text{pert}} \approx 10 \log \left( \frac{S_{ll}^R}{[S_{11}^R]} \right) \]  
(14)

when \( r_c \ll \frac{1}{4(\beta_2 - \beta_1)} \). Substituting (6) into (14), we get
\[ \{\Delta R_{ll}\}^{\text{pert}} = 10 \log \left( \frac{I_{ll}}{I_{11}} \right)_{\text{pert}} \approx 10 \log \left( \frac{\sin^2(\frac{P}{2})}{\sin^2(\frac{P}{2} \theta_1)} \right). \]  
(15)
For Lambert’s law scattering,
\[ [S^R_{mn}]_{Lam} = \mu \sin^{1/2}(\theta_m) \sin^{1/2}(\theta_n), \]  
(16)

the difference between the reverberation levels is
\[ \{\Delta R_{12}\}^\text{Lam} \triangleq 10 \log \left( \frac{I_{22}}{I_{11}} \right) \approx 10 \log \left( \frac{\sin^2(\theta_2)}{\sin^2(\theta_1)} \right). \]  
(17)

3. SINGLE-MODE REVERBERATION EXPERIMENTS AND MODEL-DATA COMPARISON

The single-mode reverberation experiments were carried out at three sites in the China Sea. One was carried out in the south sea of China in June 2012 at 22.105° N, 115.511°E (designated site C in this paper). The other two were in the Yellow Sea in August 2011 at 35.592°N, 121.065°E (designated site A in this paper), and in December 2011 at 36.065°N, 121.328°E (designated site B in this paper). The source was a low-frequency VSA. The center frequency of the transducer was 700 Hz, and its 3 dB band was 600 to 900 Hz. The acoustic signal was received by a VRA. The profile at site B was a typical profile of winter conditions, and the profiles at other two sites were the typical thermocline profiles of summer conditions. A CW pulse signal was used in the experiment.

![Fig.1 Reverberation level decay curves at site A](image1)

**Fig.1** Reverberation level decay curves at site A

![Fig.2 Reverberation level decay curves at site B](image2)

**Fig.2** Reverberation level decay curves at site B

![Fig.3 Reverberation level decay curves at site C](image3)

**Fig.3** Reverberation level decay curves at site C
The reverberation level (normalized by the electric power of the transmitted signal) decay curves of the single-modes are shown in Figs 1-3. The curves represent the averages over 20 reverberation data samples. The average difference of the reverberation level between the first and second modes at the close range $\Delta R_{12}^{\text{exp}}$ can be obtained from these curves.

The values of $\Delta R_{12}^{\text{exp}}$ are listed in Table 1. For comparison, the model predicted values given by Eqs. (15) and (16) are also listed in the Table 1. Comparing with the experimental results in Table 1, we can see that at site A and B, the single-mode fields were excited well, $\Delta R_{12}^{\text{pert}}$ is very close to $\Delta R_{12}^{\text{exp}}$, and $\Delta R_{12}^{\text{Lam}}$ is lower by about 5 dB. At site C, the single-mode field was not excited well and $\Delta R_{12}^{\text{exp}}$ is not as well described by $\Delta R_{12}^{\text{pert}}$, but $\Delta R_{12}^{\text{exp}}$ is greater by 2.8 dB than $\Delta R_{12}^{\text{Lam}}$.

<table>
<thead>
<tr>
<th>Experimental site</th>
<th>$\Delta R_{12}^{\text{exp}}$</th>
<th>$\Delta R_{12}^{\text{pert}}$</th>
<th>$\Delta R_{12}^{\text{Lam}}$</th>
</tr>
</thead>
<tbody>
<tr>
<td>Site A</td>
<td>11.0</td>
<td>11.5</td>
<td>5.7</td>
</tr>
<tr>
<td>Site B</td>
<td>11.0</td>
<td>11.8</td>
<td>5.9</td>
</tr>
<tr>
<td>Site C</td>
<td>8.0</td>
<td>10.4</td>
<td>5.2</td>
</tr>
</tbody>
</table>

Table 1 difference of the reverberation level between the first mode and second modes

4. CONCLUSION

The characteristics of the BSM at low grazing angles were obtained from three single-mode reverberation experiments in three shallow-water waveguides at different sites with different water depth and water sound speed profiles. In the experiments, the first and second single-mode fields were excited successfully by the optimum closed-loop control method and then the corresponding low-grazing-angle reverberation data were acquired. The model-data comparison shows that at low grazing angles, the BSM based on Bass perturbation theory is consistent with the experimental data, but the BSM based on Lambert’s law is far from the experimental results.

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OBSERVE SEISMIC ACTIVITIES AND AMBIENT NOISE OF UNDERWATER ACOUSTIC DATA FROM MACHO HYDROPHONE

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Abstract: Since Taiwan is located in the Circum-Pacific Seismic Zone, there are thousands of earthquakes which had been detected within one year. In addition, tsunamis, which were caused by earthquakes, have been suffered parts of areas in Asia-Pacific recently years. For the reason that Central Weather Bureau (R.O.C.) has built the first submarine cabled observatory to investigate earthquake, tsunami and ocean environment for monitoring the natural disasters at the northeastern of Taiwan’s offshore in 2011 - Marine Cable Hosted Observatory (MACHO) system. The National Science Council of the Republic of China has also sponsored a project “Marine Observatory in the Northeastern Taiwan (MONET)” to efficiently analyze the large quantity of long-time monitoring raw data from hydrophone sensors in the MACHO system. The detector in this paper includes time-varying ambient noise level estimation via Leq (equivalent continuous sound level, averaged over 30 seconds) and estimation theory, and an energy detector with the estimated ambient noise level as threshold. Two years of data are analyzed with the detector, and the results are showed seismic activities of local environment, the seasonal variation of the local ambient noise level in MACHO system and the efficiency of analysis of large amount of data. In addition to this, a user friendly UI (user interface) is written to enhance its utilization as an useful intermediary for Central Weather Bureau detecting earthquakes by the underwater acoustics. (This work is sponsored by National Science Council (R.O.C.), under Project entitled “Physical Oceanography and Acoustic Applications at the Marine Observatory at the Northeastern Taiwan (MONET) coordination project and subproject 1: Study of Underwater Acoustic Signature Extraction and its Data Base Automation” Project No. NSC 101-2221-E-002 - 028 -MY2).

Keywords: MACHO, MONET, Kalmen filter, earthquake, Parallel Processing, Hadoop Distributed File System
1. INTRODUCTION

The famous natural disaster in the early 21st century, Asian Tsunami in 2004 and Miyagi earthquake and tsunami in Japan, 2011, have shocked lots of Asian countries to notice this issue. Because Taiwan located in the Circum-Pacific Seismic Zone, Taiwan’ government has noticed this issue and operated the first underwater observatory, Marine Cable Hosted Observatory (MACHO) system in September, 2011. [1-3]. That cable is about 45 km far away from coastline of Tou-chen and the underwater node has different scientific devices, such as broadband seismometer, acceleration seismometer, tsunami pressure gauge, hydrophone and conductivity, temperature, depth (CTD) sensors. This paper is focus on the data from the single hydrophone and CTD sensor.

Hydrophones have been used to detect seismic signals and the crucial tsunami-genesis since 1950s. In middle of 20th century, Ewing’s groups found the SOund Fixing and Ranging (SOFAR) channel in the world’s ocean [4, 5] and also discovered the “T-phase” signals from hydrophone array. The applications of T-phase could be used to estimate the length of the fractured region, develop on tsunami warnings. Slack (1999) had also pointed out advantages for the use of hydrophone, for examples, detecting seismic wave, analysing regional seismic and the oceanic environment, locating the epicentre, knowing more about the lower crust and upper mantle of oceanic plates [6]. Studies showed that if the earthquake was too small to be recorded by land-based seismometers, it was more easily recorded by hydrophones to detect the T-phase signals of low-magnitude seismicity [7]. Because large seismic moments have relatively more energy at higher frequencies in their T phases [8]. Comparing with broadband seismometer, hydrophones have higher information in frequency domain, which will surpass 20 Hz, therefore, hydrophones are more easier to observe even at great distances [8]. Reviewing literature said that the hydrophones could detect the microseismicity (mb ≤ 3.5) from T waves; thence, supplying for land-based seismometers or at where were the ocean bottom seismograph (OBS) alone [9, 10].

However, in presently we are limited on locating the epicentre, because we just have one hydrophone in MACHO system. But we have the ability to analyze those passive acoustic monitoring data in time series and spectrogram from hydrophone, and observe the earthquake phenomenon and ambient noise in local environment. Therefore, we also wrote a friendly user interface, for efficiently processing that increasing years by years’ high frequency acoustic recordings (about 1T in a week). In next section, we will explain how we process those acoustic data, and in the third section use computer science technique to set a UI system. The final part discusses the different seasonal ambient noise and earthquake events which output from MACHO system.

2. METHODOLOGY

The type of hydrophone in MACHO system is RESON TC4032, and its sampling rate is up to 192 k Hz. That set two gains ─ 0 dB and 20 dB, and output the wav format for 30 seconds. Because the high sampling rate, the quantity of each day is over 120 GB, and each month is over 3.7 TB. Therefore, how to efficiently and automatically process those long time monitoring data is very necessary.

The first step is “analysing in spectrogram”, using the Short Time Fourier Transform (STFT), calculating the sound pressure level (SPL) by the output from “spectrogram” (nfft
The second step is “averaging” to get Leq (equivalent continuous sound level, the minimum averaged over 30 seconds). Those derivation of the formula can see Fang (2013) [11].

There we will start from how to use the estimation theory as the threshold to detection. Due to efficiently supporting estimations of past, present, and even future states, we will use the Kalman filter to estimate the next duration’s ambient noise level as the present threshold. The Kalman filter not only is an efficient computational (recursive) solution of the least-squares method, but also can do even when the precise nature of the model system is unknown. It has two parts, one is “Predict” part updated with time, see Formula. (1)-(2), which put in the predict value from recursive way to reduce the error covariance. The other is “Correct” part updated with measurement, and the goal is reducing the error to find K value. Using the priori state estimates the posteriori state by recursive steps, see Formula. (3)-(5).

A. Time Update (“Predict”)
(1) Project the state ahead
\[ \hat{x}_{k+1} = A_k \hat{x}_k + B u_k \]  
\( \hat{x}_k \) is a posterior state estimate 
\( \hat{x}_{k+1} \) is the state at \( k + 1 \) 
\( A_k \) is the \( n \times n \) matrix 
\( B \) is the \( n \times 1 \) matrix 
\( u_k \) is the control input
(2) Project the error covariance ahead
\[ P^{-}_{k+1} = A_k P_k A_k^T + Q_k \]  
\( P^{-}_{k+1} \) is the priori estimate error covariance at \( k + 1 \) 
\( P_k \) is the posterior estimate error covariance at \( k \) 
\( Q_k \) is the process noise

B. Measurement Update (“Correct”)
(1) Compute the Kalman gain
\[ K_k = P^{-}_{k} H_k^T (H_k P_k H_k^T + R_k)^{-1} \]  
\( K_k \) is the Kalman gain 
\( P^{-}_{k} \) is the priori estimate error covariance 
\( H_k \) is the Jacobian matrix 
\( R_k \) is the measurement error covariance matrix
(2) Update estimate with measurement \( z_k \)
\[ \hat{x}_k = \hat{x}_{k}^- + K (z_k - H_k \hat{x}_{k}^-) \]  
\( \hat{x}_k \) is a posterior state estimate 
\( \hat{x}_{k}^- \) is a priori state estimate
(3) Update the error covariance
\[ P_k = (I - K_k H_k) P^{-}_{k} \]  

3. USER INTERFACE SYSTEM OVERVIEW

Multiple small earthquakes happen every day in regions around Taiwan. The majority of them occur deep within the earth's crust, so no one feels them, but MONET listening
stations might detect them nonetheless. The basic architecture is to build MONET large-scale online data retrieving engine and gather the earthquake database from CWB. Apache Hadoop is a newcomer in this computing, and is adopted as a great new software library to already existing systems. Hydrophone wav data is processed in parallel and utilize HDFS (Hadoop File System) and RAID to store the SPL. With regard to the usage, we average SPL data over a timeframe in 30 sec, 3600 sec and 86400 sec. MONET data expands quickly. Network bandwidth and limited storage space should be managed efficiently. For instance, the daily SPL data size is about 200GB; the LEQ-30 data size of a day is 50GB. We redesign the data format and index metadata to reduce I/O operations and compress data. In addition, we design a friendly user interface on web for researcher to filter the intensity distribution with respect to time or frequency in a given duration instantly. The researcher can easily get the characteristics of given conditions.\textit{(Fig 1)}

3.1 Pre-calculating and indexing data

As part of MONET, we utilized Apache Hadoop to build a fast pre-calculating and indexing system to increase the retrieving speed. In the Hadoop MapReduce programming model, computations are expressed as two user-defined functions: Map and Reduce. Map processes SPL data and generates a set of LEQ data. All intermediate LEQ data associated with the same timeframe are aggregated and received as input by the Reduce function.\textit{[Fig 2]} Managing resources usage efficiently is essential work in our project. Both SEQ data and LEQ data are stored in Hadoop Distributed File System, HDFS, which has higher flexibility over thousands of instances than traditional file system. We extract the metadata, including timeframe size, frequency and timestamps, and make the metadata searchable by using Apache Solr, a popular, blazing fast open source enterprise search platform.\textit{[12]} The data is also compressed for visualization. Considering the performance of visualization tool, web browser, we calculate LEQ data into different timeframe and display it in proper time scale. In our experience, the proper timeframes are 30secs, 3600secs, and 86400sec. Researchers can easily understand daily data distribution via our system. There is more than 250 GB data to have to be analyzed per day. In this infrastructure, we can process 250GB data in 2 hours. The most cost of our system is I/O transmission rate. Data compression is required in our project and gives us the ability to decrease disk space and storage infrastructure requirements. Fewer I/O operations need to be performed to retrieve or store the same amount of data. Therefore, for I/O-bound workloads, the query processing time can be noticeably improved.

3.2 Retrieving Data and Connecting CWB Earthquake Database

We deliver a web search interface to access hydrophone features and CWB earthquake records in given time duration. After calculating the size of the time duration which we compare relative data size, the system queries Apache Solr to get the associated file path of LEQ data. Users can directly access those LEQ data. For example, if the given time duration is a day, there is 200GB SPL data need to be retrieved at the same time. It cost a huge workload on I/O operations. Thus, we also calculate the general intensity distribution respect to time. To explore the relationship between MONET hydrophone data and CWB earthquake data, the system also search the CWB database to get precise earthquake information. In order to enhance operational efficiency, by setting, we construct an automatic load-balancing metadata search engine and backup the metadata index everyday. Built-in real-time indexing feature allows retrieval system can readily
accept the metadata update without pause Solr server. Combine these Apache projects, we are able to build large-scale retrieving engine.

### 3.3 Web Interface

Our first web application is focused on the features of hydrophone data. Earthquakes release huge energy. Due to the high frequent earthquake, we can collect sufficient dataset. In our web screenshot, it’s obvious to show the relationship between data from CWB recording device and hydrophone data. [Fig 3] By exploit this relationship, we might find undetected earthquake events. Also, we calculate the noise pressure spectrum level respect to frequency. The researchers can easily compare the spectrum level of the given time duration with the general spectrum level.

Fig 1: System architecture

Fig 2: Hadoop MapReduce Framework.

Fig 3: This is the earthquake event screenshot of MONET retrieving engine. At the right column, the upper one is the data from CWB websites; the lower one is derived from our MONET dataset.

### 4. DISCUSSION AND RESULT

#### 4.1 Variation on the spectrogram

We analyzed the recordings by MATLAB R2010a version. Seeing the Fig 4, each line in (A) part is averaging one hour’s data and the frequency range is from 5 Hz to 100 kHz. The green line is from GMT 00:00 to 06:00; the red line is from GMT 06:00 to 12:00; the orange line is from GMT 12:00 to 18:00; the blue line is from GMT 18:00 to 24:00. The (B) part is the spectrogram of that day, and we can check with (A) part’s events. From (A) part, we can see the trend of ambient noise. Firstly, there are many system noise at high frequency, the reasons are Fang (2012) explained that the aliasing effect. Because the preprocessor of underwater node did not use the anti-aliasing filter, the noise at high frequency will map on the spectrogram of low frequency as reconstructing the signal. The aliasing phenomenon leads to raise more sound pressure intensity in the low frequency range about 4 kHz to 10 kHz. The other reason is due to the system noises. The science node has other sensors beside, which may made some noises as operating, such as the constant noise from 3 kHz to 5 kHz [3, 13]. However, this paper focuses on the low frequency ambient noise below 2 kHz, noises at high frequency will not affect our analysis.
In Fig 5, we compare the weekly and seasonal variation. In (A) part, each line means one day, so we cannot see the short pulse signals, such as earthquakes in that averaging data. Compare the trend of two lines. The red line is higher than the blue line 10 dB ref 1 µPa which is below 200 Hz. From the spectrogram of (B) and (C), we can see the obvious voiceprint of ships in (C). It means some ships crossed the MACHO observatory and made big noise duration 2012/01/16-01/23. From this case, we can know that the long term averaging data can help us know more about the local environment and will be a way to monitor the local ocean.

![Fig 4: (A) Ambient noise spectrogram of MACHO hydrophone at 2011/11/09 (each line is one hour’s data. Each block in y axis is 20 dB ref 1 µPa); (B) Spectrogram from 1 Hz to 100 kHz at 2011/11/09.](image)

![Fig 5: (A) The spectral trend below 1000 Hz of seasonal and weekly ambient noise variation (Each line is made by one day. Each block in y axis is 20 dB ref 1 µPa. Red: 2012/01/16-01/23; Blue: 2011/10/16-10/23); (B) Spectrogram of MACHO hydrophone at 2011/10/16-10/23 (Below 1000 Hz); (C) Spectrogram of MACHO hydrophone at 2012/01/16-01/23 (Below 1000 Hz);](image)

4.2 Discuss the earthquake phenomenon and activities

We show the two cases of earthquakes, the one (Fig 6) is how far and the other (Fig 7) is how weak we can record and detect in MACHO hydrophone.

Fig 6 is an earthquake measuring 2.2 and 62nm far from MACHO observatory in 2012/12/09-08:28. The epicentre is at continent. In Right side, we can see obvious pulse between 120 s to 150s, it means the hydrophone records the signal which delays some seconds. Moreover, in the spectrogram also can observe the broadband signal, and that is like the paper said that “Earthquakes with large seismic moments have relatively more energy at higher frequencies in their T phases than earthquakes with smaller moments. These frequencies, often exceeding 20 Hz, are easily observed by hydrophones even at great distances.” [8] It means the hydrophone could be an independent and useful tool for monitoring the ocean earthquakes.

Fig 7 is an earthquake measuring 5.4 and 160 nm far from MACHO observatory in 2012/10/25-10:31. The epicentre is at ocean. Since the T-phase coda will produce the secondary radiators of energy or product strong forerunners or secondary arrivals, when transmitting across the islands or seamounts. For the reason, the voltage of earthquake in Fig 7 is just higher than the ambient noise 0.02, and 160 nm far from MACHO observatory, but the hydrophone still can detect the earthquake signal.
Finally, we collect the time list of earthquakes from 2011/10-2013/04 (information from C.W.B.) and use the interface to detect how many earthquakes we can detect in MACHO hydrophone. The Table1 shows that if the epicentres are at Ocean and the percentage of correct detection is 94.42%. If the the epicentres are at Continent and the percentage of correct detection is 89.47%. The all percentage of correct detection in MACHO hydrophone is 91.22%. It shows the high performance in detecting. The reasons of false alarm are the distance of the epicentres is too far from MACHO observatory and the intensity of earthquake is too small.

<table>
<thead>
<tr>
<th>The epicentre</th>
<th>No. of earthquake</th>
<th>No. of false alarm</th>
<th>Percentage of correct detection</th>
</tr>
</thead>
<tbody>
<tr>
<td>Ocean</td>
<td>269</td>
<td>15</td>
<td>94.42%</td>
</tr>
<tr>
<td>Continent</td>
<td>494</td>
<td>52</td>
<td>89.47%</td>
</tr>
<tr>
<td>All</td>
<td>763</td>
<td>67</td>
<td>91.22%</td>
</tr>
</tbody>
</table>

5. CONCLUSION

From the above results, the hydrophone shows its ability to detect the microseismicity (mb ≤2) far from the MACHO observatory about 65 nm. Moreover, it can detect the farthest distance is 160 nm from the observatory. Since the bathymetry can strongly influence the character of a T phase. Therefore, the percentage of correct detection will also affect by the source of epicentre. Moreover, if the transmission path has bathymetric features, such as island or seamount chains, it will affect the receiving time, amplitude and duration of T phase.

In this research, we also provide an Apache Hadoop system and open source project to manage the MONET data and CWB earthquake real-time data. Meanwhile, we build a retrieving engine to make researcher filter the data by data, frequency or intensity. With these infrastructures and friendly interface, researchers can efficiently access the already existing database, in solving related scientific and academic problems. This user interface can
not only provide faster searching for mining the MACHO data, but also could be an educational platform for knowing more about earthquakes, tsunami or marine mammal’s vocalization for different users.

6. ACKNOWLEDGEMENTS

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REFERENCES


AN OVERVIEW OF OCEAN AMBIENT NOISE AROUND TAIWAN: MEASUREMENT AND ANALYSIS

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Abstract: Ocean ambient noise, the important parameter in sonar application, includes diverse sources like waves, ships, and marine life, etc. On the other hand, ocean environment (bathymetry, sediment, and sound speed profiler) which plays an important role in ocean ambient noise is complex and various especially in Taiwan. It is only 100 meter deep in the Taiwan Strait, but over 3,000 meter in the eastern Taiwan. Moreover, the warm and saline Kuroshio Current passes through eastern Taiwan, and the intrusion of its branch occurs in the northern and southern Taiwan. Since 2005, many underwater recording systems are deployed in different season and location around Taiwan. Above-mentioned systems include Several Hydrophone Recording Unit (SHRU), SM2M marine recorder, and Passive Aquatic Listener (PAL), etc. The numerous ambient noise acoustic data contains irregular transient signals like strumming noise and sonar. These transient signals should be removed to improve the data quality. In this study, a simple method is applied for removing strumming noise and sonar signals. The characteristics of the ocean ambient noise can be quantified using statistical methods to know the daily, monthly, seasonal difference in each location. Furthermore, Features of ocean ambient noise in different locations are compared and analyzed to understand the spatial changes of ocean ambient noise. The result of this study provides the idea how the ocean ambient noise changes around Taiwan for scientific or military application.

Keywords: underwater, ambient noise, Taiwan, noise removal
1. INTRODUCTION

Ocean ambient noise, an important parameter in sonar application, includes diverse sources like waves, ships, and marine life, etc[1]. On the other hand, ocean environment (bathymetry, sediment, and sound speed profiler) which plays an important role in ocean ambient noise[2] is complex and various especially in Taiwan. For example, it is only 100 meter deep in the Taiwan Strait, but over 3,000 meter in the eastern Taiwan. Moreover, the warm and saline Kuroshio Current passes through eastern Taiwan[3], and the intrusion of its branch occurs in the northern[4] and southern Taiwan[5]. Since 2005, many underwater recording systems are deployed in different season and location around Taiwan. These systems include Several Hydrophone Recording Unit (SHRU), SM2M marine recorder, and Passive Aquatic Listener (PAL), etc. In this study, acoustic data are measured and analysed to understand the overview of ambient noise around Taiwan.

2. MEASUREMENT

Acoustic measured data from PAL, SHRU, and SM2M are generally used in this study. The specification of each system is described as below:

(1) PAL: The passive acoustic listener is a low-noise, wide-band, low-duty cycle underwater recorder. The nominal sensitivity of these instruments is -160 dB re 1 V/µPa. Band-pass filters are present to reduce saturation from low frequency sound (high pass at 300 Hz) and aliasing from above 50 kHz (low pass at 40 kHz) [6]. A data sample consists of a 4.5 s time series at a sampling rate of 100 kHz. Each of these time series is fast Fourier transformed (FFT) and spectrally compressed with frequency resolution of 200 Hz from 100 to 3000 Hz and 1 kHz from 3 to 50 kHz[7].

(2) SHRU: The Several Hydrophone Receiving Unit was designed as a marine acoustic or geophysical data logger with capable of 30 day or longer deployments depending on the number of channels, data rate and the amount of data. The SHRU is 1, 2, or 4 channel data logger and the highest selectable sample rate is 19531.25 Hz (1 or 2 channel only) or 9765.625 Hz (4 channel data). Its hydrophone sensitivity typically used -170 dB re 1 V/µPa and the fixed gain is 6 or 26dB [8].

(3)SM2M: The Song Meter SM2M+ Submersible was design by Wildlife Acoustics. The sample rate is up to 48 kHz, sensitivity is -165dB re 1 V/µPa, frequency range of hydrophone is 2 to 48,000 Hz, the selectable gain are 0 and 12 dB, and it can record up to 43 days with alkaline D cell batteries.

3. MEASURED DATA

Acoustic data in this study were measured in 2006, 2007, 2009, 2012, and 2013. These measurements were located in the north-eastern, south-western, and western coastal in Taiwan. Different from PAL data with fixed frequency resolution, SHRU and SM2M provides time series data of voltage. Thus, the data analysis can be more flexible. In this study, the noise level of each SHRU and SM2M spectrogram was calculated using short-time Fourier transform with 1 Hz band, 1-second Hamming windows, and 50% overlapping. In addition, 1/3 octave band analysis is applied for data comparison. The following are the measured data of waters around Taiwan.
3.1 North-Eastern Taiwan

There were several measurements in the waters of north-eastern Taiwan (Fig. 1). In 2007, the PAL was deployed in the Mien-Hua submarine canyon for 3 months (Fig. 2). In 2009, 3 SHRUs were deployed in the North Mien-Hua submarine canyon for 1 week to 1 month (Fig. 3, Fig. 4, and Fig. 5). However, there are many cable strumming noise in these three SHRU data. In this study, a cable strumming noise removal method was applied. This method will be introduced in next chapter. In 2012, 1 SHRU was deployed on the Ryukyu Arc near the Turtle Island for 1 month (Fig. 6).

Fig. 1: Bathymetric map showing the location of the measurements in the north-eastern Taiwan.

Fig. 2: Spectrogram of the PAL data in the Mien-Hua submarine canyon in Jul.-Nov. 2007.

Fig. 3: Spectrogram of the SHRU data in the North Mien-Hua submarine canyon in Aug.2009.

Fig. 4: Spectrogram of the SHRU data in the North Mien-Hua submarine canyon in Sep.2009.
3.2 South-Western Taiwan

There were 3 sites for measurements in the waters of south-western Taiwan (Fig. 7). In 2006, the PAL was deployed near the Liouciou Islet. The PAL data in 2006 here is about 6 months (Fig. 8). In 2012, 1 SHRU was deployed on the Hengchun Ridge for 1 month (Fig. 9). In 2013, 2 SHRUs were deployed near the Fangliao submarine canyon for 1 month in each measurement (Fig. 10 and Fig. 11).
3.3 Western Coastal

There were 3 sites for measurements in the western coastal of Taiwan (Fig. 12). In 2013, the 3 SM2Ms were deployed in the Changhua coastal. There are periodic and strong croakers’ calls and snapping shrimp noises at 1 kHz in the midnight in these measurement sites (Fig. 13, Fig. 14, and Fig. 15). In general, the noise level is higher than other deep waters.

Fig. 12: Bathymetric map showing the location of the measurements in the western coastal of Taiwan.
4. DISCUSSION

The level of ambient noise around Taiwan is inversely proportional to the depth of deployed location. However, there is high-level noise in the midnight in the western coastal (Fig. 16) because there are many croakers and snapping shrimps in this region. This periodic noise makes the variation of ambient noise extremely high (up to 40 dB at 1 kHz). Besides, the cable strumming noise significantly decreases the data quality, especially the SHRUs data measured in the North Mien-Hua submarine canyon in 2009. In this chapter, a simple method is provided to remove cable strumming noise. The flowchart of cable strumming removal method is shown in Fig. 17: Flowchart of cable strumming noise removal method. In this study, the number of manually chosen reference noise level is 10 (N=10) and the threshold is 5 dB above averaging NLref (M=5). In this method, the level of threshold is inversely proportional to the number of removal. In other word, the method with lower threshold (smaller M) removes more transient high-level noise like cable strumming noise. In this study, when this method with M=5 is applied, not only cable strumming noise but other strong transient signals like sonar can be quickly and effectively removed (Fig. 18).
5. SUMMARY

This study shows more than 10 datasets of ambient noise around Taiwan. It provides the concept of ambient noise around Taiwan. Besides, a cable strumming noise removal method is introduced to improve acoustic data quality.
6. ACKNOWLEDGEMENTS

This work was sponsored by Ministry of Science and Technology, Taiwan. We are grateful to Dr. Barry Ma for the PAL, Prof. Wen-der Liang for the SHRU, Mr. Wen-Hwa Her for the mooring preparation, and the captain and crew members of the *R/V Ocean Researcher I, II, and III* for their contribution to the success of these experiments.

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Organizers: Charalampos Tsimenidis and Oliver Hinton
HIGH RATE UPLINK ACOUSTIC COMMUNICATION FROM AUVS TO SURFACE PLATFORM

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Abstract: Autonomous underwater vehicles (AUVs) are widely used in ocean operations. They often require a robust and high data rate communication link in order to transmit the large amount of survey data back to the surface platform. Those requirements face the problem of inter symbol interference (ISI) caused by the multipath reflections. Methods used to alleviate the effects of the ISI include linear or decision feedback equalization. However, these techniques require a high signal to noise ratio (SNR) and a large number of equalization filter taps needed to cover the length of the acoustic channel’s impulse response. Both of those conditions are restrictive. A new receiver structure that employs a vertical array with narrow steerable beam that suppresses multipath has been proposed in the paper. The received SNR is improved by the array processing, resulting in enhanced communication reliability. The acoustic narrow radiation beam pattern of the array adaptively tracks the direct path channel in order to suppress the multipath signals that are the main causes of ISI. This makes the follow-up channel equalization easier and more effective due to shorter impulse response of the channel. The simulation results show that the proposed uplink acoustic communication technique has a higher data rate and reliability compared with the common receiver structure under the same channel conditions. At the same time the channel equalization computation complexity is reduced by at least half. The results attest to the robustness and practicality of the proposed uplink acoustic communication scheme.

Keywords: Channel equalization, Underwater acoustic array, Beam-forming and steering underwater acoustic communication, Autonomous underwater vehicles
1. INTRODUCTION

Autonomous underwater vehicles (AUVs) are often used in ocean operations such as pipeline surveys, harbor security and others. Usually an acoustic modem that is installed on the AUV is only used to transmit the command confirmation or other short data such as its position or status [1]. The massive survey data is usually stored on the disk and the control center has to wait for the AUV to return to the surface to access the survey data. However some survey missions such as harbor security need instant high data rate communication and instructions. This capability will greatly enhance the survey missions. The high acoustic communication rate faces the problems of noise and multi-path channel especially in long distance shallow water communication [2]. In recent years, many receiver structures have been proposed to overcome those problems [3-5]. The channel equalization technique, such as linear equalization or LE [6] and decision feedback equalization or DFE is used [7]. Both of these techniques face the problems of long and time-varying channel impulse response. In the paper a new structure of the receiver which employs a vertical array has been proposed. The acoustic narrow radiation beam pattern of the array adaptively tracks the maximum path in order to suppress the ambient noise and the multipath signals that are the main causes of ISI. This makes the follow-up channel equalization easier, resulting in enhanced communication reliability. The paper consists of four parts: the channel and array signal model, the receiver structure, the simulation section, and the conclusions.

2. CHANNEL MODEL

Because of the ocean surface waves and the relative motions between the two communicating points, the underwater acoustic channel between the AUV and surface platform is a time-varying multi-path channel which is modeled as [8].

\[ h(t, \tau) = \sum_{k=1}^{K} A_k(t) \delta[(1 + \Delta_k(t)) t - \tau_k(t)] \]  

(1)

Where \( h(t, \tau) \) is the multi-path channel impulse response, \( K \) is the number of multi-path, \( k \) is subscript of each multi-path, \( A_k(t) \) is the amplitude of the \( k^{th} \) multi-path, \( \tau_k(t) \) is time delay of \( k^{th} \) multi-path. Each path has a different arrival angle at the receiver due to a different geometry and has certain Doppler frequency shifts. In order to improve the up-link communication quality, the properly oriented AUV will be almost stationary so that the Doppler effects can be neglected. It is assumed that the communication signal frame would be sufficiently short so that the channel impulse response can be considered as constant during the frame. In this sense the channel can be regarded as the static channel in each signaling frame’s duration and can be modeled as:

\[ h(t) = \sum_{k=1}^{K} A_k \delta(t - \tau_k) \]  

(2)
3. THE RECEIVING ARRAY MODEL AND STRUCTURE

3.1. Receiving Array

The ocean can be modeled as a layered medium which has boundaries in the vertical direction but a free boundary condition in the horizontal direction. A M-elements vertical linear array has been employed from the surface platform (the base) with element spacing equals to the half wavelength as illustrated in Fig. 1. In order to achieve a higher data rate the carrier signal has a relatively high frequency, and therefore the array has a small size. It is also assumed that the underwater acoustic channel (UAC) between the AUV and all array elements is the same.

\[ r_m(t) = s(t - \frac{(m-1)d \cos(\theta)}{c}) * h(t) + n(t) \]

As shown in the Fig.1 (b), the sound path difference between the adjacent two elements is \( d \cdot \cos(\theta)/c \), where \( c \) is the sound speed, \( d \) is the element spacing and “*” denotes convolution and \( n(t) \) ambient noise. The first element \((m=1)\) is the reference element.

3.2. Receiver structure

The scenario of the underwater acoustic communication from the AUV to the surface platform is as shown in Fig.1 (a). The acoustic narrow radiation beam pattern of the receiving array points to the direct path of the multi-path. The multipath signals reflected by the surface and bottom are suppressed. The receiver structure is shown in Fig.2. The signal frame structure is adapted, and the receiver processes the received signal one by one frame. First the synchronizer detects the synchronization signal that is located at the beginning of each frame. The synchronizer controls the beam-former to detect the
incidence angle of the direct path once the sync signal is detected. Then the receiver delay 
stack the received array signals to form the beam pattern and steer it at the angle of the 
direct path. The received in-beam signal is then demodulated to reconstruct the transmitted 
symbols.

![Fig.2: Diagram of receiver structure](image)

4. SIMULATION STUDY

In order to verify the system performance, the communication in the different channels 
A and B has been simulated. Both simulations have the same system parameters shown in 
Table 1 and use QPSK modulation and constant sound speed c=1500 m/s.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Carrier frequency / Bandwidth</td>
<td>18kHz / 6kHz</td>
</tr>
<tr>
<td>Signal to noise ratio</td>
<td>10dB</td>
</tr>
<tr>
<td>Symbol / Frame duration</td>
<td>0.167ms / 2s</td>
</tr>
<tr>
<td>Array element number and spacing</td>
<td>9/6cm</td>
</tr>
<tr>
<td>Number of weights in LE filter</td>
<td>10</td>
</tr>
<tr>
<td>Adaptive algorithm of LE</td>
<td>LMS and its step size</td>
</tr>
<tr>
<td>Adaptive algorithm of DFE</td>
<td>RLS and its forgetting factor</td>
</tr>
</tbody>
</table>

Table 1: The system parameters

4.1. The simulation A

The channel consists of three paths including the direct-path, the smooth reflecting 
surface, and the smooth reflecting bottom. Two typical situations are assumed. The first is 
deep-ocean and short horizontal range channel. The second is shallow-water and long 
horizontal range channel.

<table>
<thead>
<tr>
<th>Assumed Geometry</th>
<th>The 1st situation</th>
<th>The 2nd situation</th>
</tr>
</thead>
<tbody>
<tr>
<td>Array depth</td>
<td>10m</td>
<td>10m</td>
</tr>
<tr>
<td>Ocean depth</td>
<td>2000m</td>
<td>100m</td>
</tr>
<tr>
<td>Transmitter depth</td>
<td>1970m</td>
<td>70m</td>
</tr>
<tr>
<td>Horizontal distance</td>
<td>200m</td>
<td>2000m</td>
</tr>
</tbody>
</table>

Table 2: The environment parameters
The channel impulse responses for two situations are shown in Fig.4. Shown in Fig.4 (a) is the deep-ocean and short horizontal range channel. The shallow-water and long horizontal distance channel is shown in Fig.4 (b), which has the similar incidence angle in different paths. Fig.5 presents the constellation diagrams of the communication through the channel from Fig.4 (a). Comparing Fig.5 (a) with Fig.5 (d), we can see that the simultaneous beam steering and equalization that is space filter channel equalization (LE or DFE) technique proposed in this paper equalizes the multipath channel effectively. Contrasting the Fig.5 (b)(c) with Fig.5 (e)(f), the space filter channel equalization techniques (LE and DFE) support this conclusion.
Fig. 5: Constellation diagrams: (a) without any channel equalization. (b) with LE. (c) with DFE. (d) with space filter equalization. (e) with space filter and LE. (f) with space filter and DFE.

Fig. 6: Constellation diagrams (The figures are arranged in the same order with the Fig. 5.)

Fig. 6 is the constellation diagram of the communication through the channel shown in Fig. 6 (b). Comparing Fig. 6 with Fig. 5, the channel equalization technique proposed in the paper is inefficient in this situation because of the different paths having similar incidence angle. However, space filter channel equalization is again effective as shown in Fig. 6 (e)(f).

4.2. The simulation B

The channel impulse response is computed with ray model in the simulation B. As shown in Fig. 7 is the sound speed profile which was measured in a lake during the summer. The communication environment parameters are: array depth 10m, lake depth 100m, transducer depth 70m, communication range 5000m. The computed channel parameters shown in Table 3 correspond with the channel impulse response of Fig. 8.
In order to verify the system performance in the actual environment, the channel impulse response was computed with the ray model according to the actual environment parameters. The channel taken in simulation B is more complex compared with the channels in simulation A. Comparing Fig. 9 (a)(b)(c) with Fig. 9 (d)(e)(f), it can be seen that both the spatial array filtering in combination with channel equalization are effective in the assumed conditions. The 12kbp/s data rate was achieved using the single carrier with zero errors. The required bandwidth was only $B = 6\text{kHz}$.
5. CONCLUSIONS

A new structure receiver that employs a short vertical linear array has been proposed in the paper. The acoustic narrow radiation beam pattern of the array adaptively tracks the maximum path channel (direct path) in order to suppress the multipath signals that are the main causes of ISI. The simulation results show that the space filter channel equalization technique makes the follow-up channel equalization more effective, and the proposed uplink acoustic communication technique has a higher data rate and reliability compared with the common structure receiver under the same channel conditions. At the same time, the follow-up channel equalization computation complexity is reduced by at least half. The different channels were successfully tested by simulations that suggest the new structure is suitable for the general environment.

6. ACKNOWLEDGEMENTS

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REFERENCES

BUILT-IN TOOLS FOR CLOCK SYNCHRONIZATION IN UNDERWATER ACOUSTIC NETWORKS DURING PAYLOAD DATA EXCHANGE

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Abstract: Modern underwater acoustic modems, besides receiving and transmitting data, can measure signal propagation time and, evaluate the distance between the transmitter and the receiver. Such capabilities can be exploited for synchronization of underwater acoustic sensor networks, where sensing and actuation must be coordinated across multiple nodes. However, such problem as the accuracy of propagation time measurement, especially in reverberant environments, as well as the evaluation of clock skew on acoustically interacting modems, have been investigated rather poorly. One of the objectives of this paper consists in experimental evaluation of stability/precision of propagation time measurements, particularly between underwater acoustic modems communicating in reverberant environments, as well as in demonstration of the modems built-in tools providing opportunities for precise estimation of the modems clock skews. Another objective is to demonstrate a “comfortable” way for clock synchronization of underwater acoustic networks nodes directly during payload data exchange. Experiments were carried out during CommsNet13 sea trials, CMRE, La Spezia, in conditions of reverberant underwater acoustic channels on several modem pairs and at several mutual positions.

Keywords: underwater acoustic network, underwater acoustic modem, underwater communication, underwater acoustic clock synchronisation, underwater telemetry, underwater acoustic positioning
1. INTRODUCTION

Synchronizing the nodes of an underwater acoustic communication network is a necessary prerequisite for them to jointly use the sensor data they collect. For example, a coherent operation of spatially distributed underwater acoustic instruments is only possible if the timing of transmitted and received acoustic signals is precisely coordinated. Synchronization can be achieved by assessing and correcting the time offsets and skews between internal clocks of the network nodes regarding to a reference “base” clock.

The time offsets between internal clocks occur as the nodes of the communication network are not turned on simultaneously, while clock skews between internal clocks are caused by their quartz oscillators operating at slightly different frequencies.

Several sophisticated time synchronization protocols exist for terrestrial networks allowing to synchronize the internal clocks of the network nodes. A good example is the general-purpose NTP protocol (Network Time Protocol) [1] that synchronizes computers in a network. Its disadvantages, though, are its relatively low synchronization accuracy (an order of milliseconds), and the large amount of overhead data to be transmitted. Besides NTP, popular terrestrial-specific protocols include, for example, PTP (Precision Time Protocol) [2], RBS (Reference Broadcast Synchronization) [3] [4], TPSN (Timing-sync Protocol for Sensor Networks) [5] and FTSP (Flooding Time Synchronization Protocol) [6]. They provide higher synchronization accuracy and have relatively small protocol overheads. Nevertheless, when applied to an underwater acoustic channel, these protocols have a significant disadvantage - they do not take into account the signal propagation time and disregard the clock skews during the synchronization procedure (in radio networks, signal propagation time and the duration of synchronization procedure are considered negligibly small).

For underwater channels of the most practical interest, propagation delays of acoustic signals are significantly larger than those of a radio signal. For example, when transmitting over a relatively small distance (hundreds of meters), the propagation delay of an acoustic signal comprises hundreds of milliseconds, whilst a radio signal delay does not exceed one microsecond.

This large difference in signal propagation delays explains the need to develop specific protocols for underwater acoustic sensor networks. Yet, development of effective synchronization protocols needs special tools allowing for accounting all propagation delays between OSI layers and for accurate measuring the propagation delays in hydro-acoustic media.

In the next sections we present such tools, implemented as special functions of S2C (Sweep-Spread-Carrier)-series underwater acoustic modems [7] version 1.7 [8]. They provide means to measure acoustic signal propagation delays with microseconds accuracy, in particular when operating in dynamic, reverberant underwater acoustic channels. As a distinct feature, these tools allow to synchronize or re-synchronize the clocks (to specify clock offsets and skews) directly during payload data exchange between underwater network nodes. Section 1 describes these tools, while section 2 provides experimental results.
2. ACOUSTIC SIGNAL PROPAGATION DELAY MEASUREMENT

The tools presented in this section are based on the extended special features of S2C-series underwater acoustic modems. These features offer the user a set of AT commands to create own synchronization algorithms for an underwater acoustic network.

It is known [9], that in digital communication one of the most significant time-of-flight (TOF) measurement errors that occur during data exchange between a transmitter and a receiver is caused by unknown duration of a packet’s transfer between the OSI application layer (or transport and network, if present) and MAC layer. To avoid ambiguities in the TOF measurements we considered the creation of special operation modes for our communication system, in which some tasks of the lower layers are delegated to upper-layers. With regard to synchronizing the data streams in the communication channel, this approach provides an access of upper-layer protocols to the clock of the lowest layer and, therefore, eliminates the problem of unknown packet exchange duration between them.

This approach is implemented in S2C-series underwater acoustic communication systems [7], [8]. Fig.1 illustrates the interaction of OSI layers of an S2C-series underwater acoustic modem in the special (so-called synchronous) mode.

![Fig.1. Layer interactions during synchronous data exchange](image)

As demonstrated in Fig.1, when transmitting data in synchronous mode, the upper layer of the communication system can specify exact intervals between data packets for the physical layer to transmit them into the medium. To ensure high accuracy of these intervals the physical layer uses its own clock. To inform the upper layer about the time of each packet’s actual transmission, the physical layer returns a notification including its clock timestamp.

Important feature of the synchronous mode is the ability to accurately determine signal propagation delays between the network nodes without interrupting payload data exchange between them. Another important feature is the ability to precisely evaluate clock skews between interacting network nodes (as described in section with experiments).
Besides scheduled transmissions, in synchronous mode a transmission of data packets can be initiated by an upper-layer protocol to be performed simply as soon as possible. To inform the upper layer about factual time of data packet’s transmission, the physical layer of the transmitting modem always sends a notification with the clock timestamp, at which it started the transmission.

On the remote side, once reception of the incoming signal’s has started, the receiver’s upper layer also receives a notification with the physical layer’s clock timestamp, in which the reception started. As a result, during data packet exchange the upper-layer protocols obtain means to accurately measure signal propagation delays in the underwater acoustic channel. As information about propagation delays is available on the upper layer, the user can create own synchronization algorithms for the network nodes.

Fig. 2 presents examples of two underwater modems’ operation in synchronous message exchange mode to facilitate the task of synchronizing their clocks during payload data exchange. Here the application layer (the PC as the data source/recipient) is connected directly to the underwater modem, whereas the application to the left is connected to the modem with address 1, and the application to the right is connected to the modem with address 2.

Commands and notifications in Fig. 2 illustrate the cross-layer connections, in particular, the dialogs between the application layer and the MAC layer during message transmissions into the acoustic channel and message receptions from it.

Fig. 2. Synchronous instant message exchange to facilitate the task of clock synchronization over the underwater acoustic channel

The application to the left initiates a synchronous instant message exchange (the upper left dialog). The first line includes the AT command (SENDIMS) for transmission of a message 4 bytes long to a remote modem with address 2, where the body of the 4-byte message is a text string “test”. The absence of a value between the last two commas means the message is to be transmitted as soon as possible. After receiving this command, the MAC responds with a command acceptance notification OK (the second line). The third line describes the MAC layer’s notification about the start of a synchronous message (ims) transmission (SENDSTART) to the remote modem with address 2, of 150000 us duration without a delay of command execution (the last field is zero). In the forth line, the MAC layer notifies about the end of the transmission (SENDEND) of a synchronous instant
message (ims) to a remote modem with address 2. Here the physical layer clock’s timestamp of the transmission start was $T_1=3000000000$ us. If needed, it is easy to calculate the timestamp of the transmission’s end by adding $T_1$ to the transmission’s duration of 150000 us.

On the remote side, during message reception from the hydro-acoustic medium the modem with address 2 (MAC layer) begins a dialog with its upper layer. The first line informs about detection of an incoming acoustic signal (RECVSTART). The second (RECVEND) notifies, that at reception start the physical layer’s clock timestamp was $T_2=4005000000$ us, the duration of the received message was 150000 us, the level of the received signal was -40dB, the signal integrity was 0.26 (see section 2 for details on the signal integrity parameter; signal integrity is given here as a normalised value, whereas the modem documentation describes non-normalised values). The last line contains a notification about the end of reception of a synchronous instant message (RECVIMS) 4 bytes long, received from a remote modem with address 1 sent to the modem with address 2, the physical layer clock timestamp of the signal reception start was 4005000000 us (also, if needed, the moment of reception end can be calculated by adding the message duration, 150000 us, to this timestamp), the level of the received signal was 40 dB, the signal integrity parameter was 0.26, the mutual radial velocity of the modems was 0.1 m/s, the contents of the 4-byte message was a text string “test”.

After generating its own notification, the application connected to the modem with address 2 initiates a synchronous message transmission back to the other side. The text to the lower right illustrated the dialog between the application layer and the MAC layer. The first line describes the command (SENDIMS) for transmission of a synchronous message 25 bytes long to the modem with address 1 scheduled for $T_3=4006000000$ us (according to the physical layer clock), where the 25-byte field contains the text “4006000000:100000:testOK”. Hereby, the data to be transmitted consists of two parts: the timestamp of the message transmission by the modem with address 2 (4006000000 us), and the delay 100000 us after reception of the previous message from the modem with address 1 ($T_3- T_2=100000$ us). After receiving the command to start transmission, the MAC layer notifies about command acceptance by sending the OK status for transmission to remote modem with address 1. The third line contains the notification about the start of a synchronous message (ims) transmission (SENDSTART) to remote modem with address 1, of 290000 us duration. Here the delay of command execution (the difference between accepting the SENDIMS command and SENDSTART - its execution by the physical layer), measured by the modem, was 720000 us. The forth line notifies about the end of the transmission (SENDEND) of a synchronous instant message (ims) to remote modem with address 1. Here the physical layer clock’s timestamp of the transmission start was $T_3=4006000000$ us.

In conclusion of the exchange cycle the modem with address 1 receives the message returned by the modem with address 2 through the acoustic channel. The MAC layer of this modem begins a dialog with the application layer (at the lower left). The first line notifies about detection of an incoming acoustic signal (RECVSTART). The second line (RECVEND) informs, that at reception start the physical layer’s clock timestamp was $T_4=3004000000$ us, the duration of the received message was 290000 us, the level of the received signal was 45 dB, the signal integrity was 0.28. The last line contains a notification about the end of reception of a synchronous instant message (RECVIMS) 18 bytes long, received from a remote modem with address 2 sent to the modem with address 1. The level of the received signal was -45 dB, the signal integrity was 0.28, the mutual radial velocity of the modems was also 0.1 m/s. The contents of the 25-byte message was
a text string “4006000000:1000000:testOK”, containing the $T_3$ value and the timestamp difference $T_3 - T_2$.

After commencing the exchange cycle the application, connected to the modem with address 1 has all the information required for synchronizing its clock with the clock of the modem with address 2. In particular, it can calculate the signal propagation time in the underwater channel as $T_p = (T_4 - T_1 - T_3 + T_2)/2 = 1500000$ us. Also, knowing the clock timestamp of the modem with address 2 allows to calculate the offset of the clock of the modem with address 1, i.e. the $t_{m1} - t_{m2}$ offset as $dT = T_4 - T_3 - T_p = -1003500000$ us.

It is obvious, that after multiple synchronous message exchanges, besides clock offsets, the clock skews can be evaluated as well.

It should be noted, that data, transmitted as the contents of a synchronous instant message is binary. Data from multiple sources can be combined, but the size of one message should not exceed 512 bits.

Also, the examples illustrated with text in Fig. 2, in reality take up much less space. In particular, due to the fact that for binary representation of a timestamp one 4-byte word is enough, what is much less than the timestamp ASCII-representation in form of mentioned above ten 1-byte words.

Hence due to the modem’s specialized mode of operation that allows to delegate several MAC layer tasks to an upper layer, it is possible for it to use the physical layer clock to control signal transmissions into the medium and register signal receptions from the underwater channel with physical layer clock as well. As a result, the ambiguity of signal transmission/reception time, caused by its transfer between the OSI layers, is eliminated.

3. EXPERIMENTAL STUDY

Experiments on synchronous instant message exchange were performed during the CommsNet13 sea trials from 9 till 20.09.2013, conducted near La Spezia in the Mediterranean Sea. The trials involved the experimental LOON (Littoral Ocean Observatory Network) – a permanent installation created by CMRE for multidisciplinary studies. The synchronous instant message exchange was performed between LOON network nodes M1 (44°02.4047’N, 09°49.8733’E, depth 27.5 m), M2 (44°01.9273’N, 09°49.6949’E, depth 28.5 m), M3 (44°02.0151’N, 09°50.3955’E, depth 27.5 m) and M4 (44°01.5458’N, 09°50.1683’E, depth 28.0 m).

The distances between corresponding pairs were: M1-M2: 912 m, M1-M3: 1001 m, M1-M4: 1634 m.

During the sea trials, the LOON was equipped with S2CR 18/34 underwater acoustic modems. The modem transducer had a smooth transmit voltage response in the operation frequency range between 18 and 34 kHz with variations up to 6 dB at the frequency band edges. The transducer had a weakly pronounced resonance at 26 kHz and was characterized with omnidirectional transmit diagram in horizontal plane and weakly expressed directivity in vertical plane (opening angle about 120°). The transmit level of the signals was 169 dB re 1 uPa/m.

An example is described below, where node M1 was transmitting synchronous instant messages in 1 second intervals, while nodes M2, M3 and M4 were receiving those messages and evaluating their clock offsets (physical layer clocks) compared to the clock of the transmitter-node M1. In addition, M2, M3 and M4 were evaluating the synchronization precision of the modem’s receiver with the incoming acoustic signal.
The clock skews were evaluated as follows. After synchronizing the receiver (physical layer) with the acoustic signal, the time elapsed since last signal reception (last signal synchronization) was estimated. In ideal case, if the clock skew is zero and the acoustic channel is reverberation-free, the time elapsed between two consequent signal synchronizations (receptions) should match the interval between those signals’ transmissions. In reality, these intervals do not exactly match. Different quartz oscillator frequencies of the transmitter and the receiver cause their clocks to drift apart. Therefore, there is always a certain difference between the transmission intervals (controlled by the transmitter’s clock) and the reception intervals (detected by the receiver’s clock). In addition, the difference between these intervals can be caused by interference introduced by the underwater acoustic channel.

At a first approximation, the clock skew is described as a linear function, defined by the difference between the clock generators’ frequencies, whereas the underwater acoustic channel’s influence is described by the random dispersion of the incoming signal registration times (synchronization of the receiver with the incoming signal).

Fig. 3 demonstrates the conditions during signal receptions, in particular, for propagation between M1 and M3. Depicted are daily monitoring results for several channel parameters measured before and after the tests.

Fig. 3a presents the multipath intensity profile, measured by the M3 node modem’s physical layer (propagation between nodes M1 and M3). On the profile’s daily monitoring results 2-4 multipath components are visible (one with excess propagation delay of about 120 us, others – about 280 us and 390 us). It should be noted, that the underwater acoustic modem only accounts for the significant-energy components, i.e. no more than 6dB weaker than the dominant component. Hence, the lower-energy components are absent on the profile illustrations. The signal level (as a sum of its component energies) exceeded the noise level by 8-17 dB (Fig. 3b). The signal integrity, Fig. 3c, was between 0.2 and 0.6 (this parameter was measured as the correlation coefficient of the synchronous component with the reference signal). The Doppler scattering of the signal was within ±2 Hz (Fig. 3d).

The clock skew, experimentally estimated after synchronous instant message exchange between nodes M1 and M3 is presented in Fig. 4a. As seen in the figure, the physical layer clock of the M3 modem runs behind the clock of the M1 modem. On the measurement interval of 392 s the clock was 1589 us late. Experimental results prove, that on first approximation, the clock skew indeed can be expressed with a linear dependency.

The scatter of incoming signal detection times is presented in Fig. 4b. It is characterized by a certain non-zero mean and random scatter around it. The mean value of ~4.05 us corresponds to the difference between generator frequencies of the physical clocks of modems M1 and M3, and defines the clock skew between them. The standard deviation of 2.87 us defines the random scatter of the signal detection time and is caused by the channel’s residual reverberation (i.e. not entirely suppressed multipaths). In fact, this scatter defines the level of influence the channel has on the signal detection accuracy (synchronization of the receiver with this signal). It is worth mentioning, that the scatter is different from Gaussian distribution, so only on first approximation the standard deviation can be used to assess the accuracy of the receiver’s synchronization when operating in complex channels with pronounced multipath. In current case, with the standard deviation at 2.87, the extreme receiver synchronization values were at -17…+11 us around the measured mean value.

The experiments with other modem pairs gave similar results. Clock skew between M1 and M2 was also almost linear being equal to 5.23 us per second (mean value). The
standard deviation from the measured mean value was 2.62 \text{us}. The extreme deviation values were at -3…+11 \text{us} around the mean. Clock skew between M1 and M4 was characterized as 1.58 \text{us} per second. The standard deviation from this value was 0.84 \text{us}. The extreme deviation values were at -2…+4 \text{us} around the mean.

Fig. 3. Signal reception conditions (propagation between nodes M1 and M3)

Detailed investigation of the channel’s influence onto accuracy of the clock skew estimation, in particular, finding a quantitative connection between the generalized parameters of the channel’s complexity and the accuracy of the receiver’s synchronization with the incoming signal, is the topic of a journal paper coming soon.

Fig. 4 Clock skew between nodes M1 and M3 (a), and the M3 signal detection time scatter, caused by channel interference (b)
4. DISCUSSION AND CONCLUSIONS

The paper presents the results of newest S2C-series underwater acoustic modems to create specialized operation modes, which allow to precisely synchronize modems’ clocks directly during payload data exchange between them. In particular, S2C-series modems were fitted with additional cross-layer connections enabling to delegate several media access control tasks to upper-layer protocols of the communication system. There has been presented the built-in tools and AT commands that provide the upper-layer protocols with information about signal transmission/reception time according to the underwater acoustic modem’s physical layer clock. So user-specific applications have an opportunity to eliminate errors caused by undefined duration of data packet transmissions between the upper-layers and the physical layer.

The use of synchronous instant messages does not require performing a time-consuming procedure of initial clock synchronization and further re-synchronizations. With synchronous instant messages, even the first synchronization’s accuracy after the first message exchange can be high enough, for example, as shown in section 2, it can have the order of microseconds. Further on, during payload data exchange, synchronous instant messages can include information about a determinate delay of response generation, as well as the current clock value of the responding modem into the data being transmitted. This information allows the receiver to sync up its clock to the responder’s one, and refresh or refine the estimation of the clock skew directly during data exchange.

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MULTIUSER COMMUNICATION BY
ADAPTIVE TIME REVERSAL IN DEEP OCEAN

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Abstract: Recently, demand for multiuser underwater acoustic communication has increased, e.g. for multiple AUVs operation. Passive time reversal is an promising approach to realize multiuser communication, because signals from differently positioned sources can be separated, while intersymbol interference (ISI) is removed, by its spatio-temporal focusing. Additionally, to enhance cancelling crosstalk, adaptive time reversal has been proposed by Kim et.al. In this paper, the effectiveness of adaptive time reversal for multiuser communication in deep ocean is discussed with experimental data and simulation using normal mode method including source movements, comparing with multiuser multichannel DFE. As results, the performance of adaptive time reversal is better than that of multiuser multichannel DFE, especially in case that sources are positioned closer.

Keywords: time reversal, underwater acoustic communication, multiuser communication, crosstalk mitigation, Doppler
1. INTRODUCTION

Recently, demands for underwater acoustic communication, multiuser communication from multiple sources has increased for communication with multiple autonomous underwater vehicles (AUVs) or underwater acoustic communication network [1,2]. Especially in Japan Agency for Marine-Earth Science and Technology (JAMSTEC), AUVs has been developed for exploration of ocean seabed resources. However, for example, in case of using code-division multiple access (CDMA) [3] similarly in radio communication, data symbols are spread with some orthogonal codes, so that throughput is sacrificed to discriminate different users.

In the meantime, time reversal is an attractive solution to achieve multiuser communication [2,4,5]. Originally, in the case of single user communication, the intersymbol interference (ISI) due to rich multipath environment inherent in the ocean acoustic propagation is the major problem for underwater acoustic communication. Spatial and temporal focusing of time reversal removes such ISI by collecting multipath signals. In other words, time reversal can turn multipath signals as obstacles into a benefit. Additionally, time reversal can be applied to multiuser communication. By its spatial focusing, signals from sources at different positions can be separated each other by the same processing in the single user case without sacrificing throughput. Furthermore, to enhance cancelling crosstalk, adaptive time reversal has been proposed by Kim et.al [6,7]. In this paper, supposing communication from multiple AUVs, the effectiveness of adaptive time reversal for multiuser communication in deep ocean is discussed with experimental data and simulation using normal mode method including source movements, comparing with multiuser multichannel DFE [8].

2. TIME REVERSAL FOR MULTIUSER COMMUNICATION

In active time reversal, which is applied to multi-input-single-output (MISO) communication, multipath waves are converged in space and time to the focus point in practice. Thus, multiple focusing is achieved in addition to removing ISI. In the meantime, in passive time reversal [9] , which is applied to single-input-multiple-output (SIMO) communication, similar spatial and temporal focusing effect can be obtained by signal processing on the receiver array side. The objective of this paper is to investigate multiuser communication from multiple sources like AUVs to a receiver array (base station). Thus, only passive time reversal is discussed hereafter.

In passive time reversal, a probe signal is transmitted from a source, followed by an information-bearing signal. The channel impulse response (CIR) from the source to the receiver array is obtained from the received probe signal, which is cross-correlated with the received information signal at each channel, and the resultant signals are summed over the channels. This process is equivalent to active time-reversal process. Thus, by spatial and temporal focusing effect, in addition to removing ISI, signals from sources at different positions can be separated.

Additionally, in order to enhance crosstalk mitigation, adaptive time reversal is applied in this study, which has been proposed by Kim et al [6,7]. The theory is explained in brief here in case of two users. Supposing CIR, \( h_j(t) \), received at the \( j \)th element of the receiver
array from the \( i \)th user (source), and its expression in the frequency domain, \( H'_i(f) \), a column vector \( \mathbf{d}_k \) is defined as

\[
\mathbf{d}_k = \left[ H'_1(f) \ldots H'_M(f) \right]^T,
\]

(1)

where \( M \) is the total number of receivers.

To further suppress the crosstalk, an adaptive time-reversal filter for user 1, \( \mathbf{w}_1 \), is given by,

\[
\mathbf{w}_1 = \frac{\mathbf{R}^{-1}\mathbf{d}_1}{\mathbf{d}_1^T\mathbf{R}^{-1}\mathbf{d}_1} \quad \text{where} \quad \mathbf{R} = \mathbf{d}_1\mathbf{d}_1^T + d_2\mathbf{d}_2^T + \sigma^2\mathbf{I},
\]

(2)

subject to the constraint that \( \mathbf{w}_1^T\mathbf{d}_2 = 0 \). Here, \( \mathbf{d}_1^* \) denotes the complex conjugate transpose and \( \sigma^2\mathbf{I} \) is a small diagonal loading for a matrix inversion with an identity matrix \( \mathbf{I} \).

Similarly an adaptive time-reversal filter for user 2, \( \mathbf{w}_2 \), can be derived in Eq. (2) by substituting \( \mathbf{d}_1 \) to \( \mathbf{d}_2 \). By calculating Eq. (2) for all frequencies in the bandwidth and converting them to time domain, the adaptive time-reversal filter, \( h'_i(t) \), can be obtained, which replaces, \( h'_i(t) \) in conventional time reversal.

In this paper, after both of adaptive and conventional time-reversal combining, a single channel decision feedback equalizer (DFE) is combined to remove residual ISI similarly as in the previous studies[5,10].

3. ANALYSIS WITH EXPERIMENTAL DATA

The experiment was carried out in Suruga-bay, as shown in Fig. 1. The source was suspended from R/V Kaiyo, and a 20-channels receiver array was installed at the point 30 km away from the source as shown in Figs. 1(a) and (c). The intervals of receives were \( \sim 6 \) m, that is, the total length of the array was \( \sim 120 \) m. The source depth was changed from \( \sim 300 \) to \( \sim 1400 \) m as shown in Fig. 1(c). Signals transmitted from different depth were synthesized to simulate multiuser communication signals, similarly as in the previous study[ ]. The depth of the receiver array was from \( \sim 840 \) to \( \sim 940 \) m, also as shown in Fig. 1(c). The sound speed profile around the source point is shown in Fig. 1 (b).

The source level was \( \sim 196 \) dB re 1\( \mu \)Pa at 1 m and the bandwidth was from 450 to 550 Hz. In this experiment, the information-bearing signal, which is 2048 symbols modulated with binary phase shift keying (BPSK) at the data rate of 100 bits/s, was transmitted. The initial 200 symbols were training symbols for DFE.

As mentioned above, the two signals transmitted from different depth were synthesized to simulate two users communication signals. Such synthesized-multiuser test signals were tried to be modulated with the following four methods, conventional time reversal with a single channel DFE, adaptive time reversal with single channel DFE, multichannel DFE for single user, and multiuser- multichannel DFE. In the cases of conventional time reversal and single user DFE, the signal processing is the same as in the case of single user. In all these four methods, a second order digital phase-locked loop (DPLL) [11] is embedded in the DFE.

Figure 2 shows result in the case that one of the source (user 1) is located at the depth of 1202 m and the other is at the depth from \( \sim 300 \) to \( \sim 1400 \) m, as indicated in the horizontal axis of the figure. In this figure, output SNR of user 1 is shown, and SU-MDFE and MU-DFE indicate the cases of single user DFE and multiuser DFE, respectively.
This result shows that adaptive time reversal is better than the other three methods wherever user 2 is located to user 1. Thus, adaptive time reversal is most effective for crosstalk mitigation. In the meantime, the improvement by multiuser multichannel DFE is observed comparing with single user DFE, however, both of them are not competitive to conventional and adaptive time reversal.

Fig.1: (a) Bathymetry of the experiment site, Suruga-bay, (b)sound speed profile, and (c) source and receiver array positions.

Fig.2: Demodulation results: Output SNR of user1.

4. ANALYSIS WITH SIMULATION

For more detail analysis including Doppler effect, simulations in deep ocean were conducted, in which the normal mode method proposed by Schmidt and Kuperman was used as previous studies [12,13]. When a source and a receiver are located at $r_c=(r_s, z_s)$ and
The pressure field with normal mode representation is given by,

$$p(\omega) = \frac{ie^{-iz/4}}{\sqrt{8}\pi \rho(z_r)} \sum_{m=1}^{M} S(\Omega_m) \Psi_m(z_r) \Psi_m(z_s) \frac{e^{i\omega(\omega)v_s}}{k_m(\omega) \left| r_r - r_s \right|}$$  \hspace{1cm} (3)

with $$\Omega_m = \omega - k_m(\omega)v_s$$,  \hspace{1cm} (4)

where $$S(\omega)$$ is the transmitted signal frequency spectrum at the angular velocity, $$\omega$$, and $$P(\omega)$$ is the frequency spectrum at the receiver. And $$k_m$$ is the horizontal wave number of the $$m$$th mode, the $$\Psi_m(z_r)$$ is the $$m$$th modal eigen-function, and $$\Omega_m$$ is the shifted frequency at each mode. This method can include the effect of receiver movement as well, however, in this paper, only source movement is considered.

Figure 3 shows the sound speed profile and the arrangements of sources and receivers. Here, the communication from two sources (users) is assumed. The source 1 (user 1) is fixed at the depth of 1300 m, that is, the axis of SOFAR channel and the depth of the source 2 (user 2) is changed from 1243 to 1357 m at the interval of 6 m. The 20-channels receiver array is spanned at the depth from 1243 to 1357 m, similarly, at the interval of 6 m. The distance between the sources and the receiver array is 1000 km. The transmitted signal is 2048 symbols modulated with BPSK at the data rate of 100 bits/s, same as in the at-sea experiment.

Figure 4 shows the result when the two sources move at the speed of 0.5 m/s. The averaged input SNR of the received signals is ~ 17 dB in this case. This result shows that adaptive time reversal can be very effective independently on the relative distance between two sources although a little degradation is observed when the source 2 is closest, that is, 3 m away to the source 1. Meanwhile, single-user DFE and multiuser DFE are inferior to adaptive time reversal regardless of the relative source positions. Particularly, when the source 2 depth is from 1279 to 1315 m, even multiuser DFE fails in demodulation. Thus, in case that two source are positioned close, adaptive time reversal has a better performance to remove multiuser interference, although a little improvement with multiuser DFE comparing to single-user DFE is observed when the source 2 depth is 1273 and 1321 m.
5. SUMMARY

Adaptive time reversal was applied to multiuser communication in deep water. In the analysis of synthesized real data, the effectiveness of crosstalk cancellation by adaptive time reversal was demonstrated. And in the analysis of simulation including source movements, it was proved that adaptive time reversal has much better performance than multiuser multichannel DFE particularly when the sources are closer.

As future work, the near-far problem and comparison with OFDM will be studied. And at-sea experiment will be carried out over several hundred kilometers range, in which the two sources transmit signals simultaneously in practice.

6. ACKNOWLEDGEMENTS

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Abstract: Application of steganography to covert underwater acoustic communications is addressed. More precisely, we propose to slightly modify previously recorded short pieces of ambient noise (biological noise, underwater works, ship noise, rain, etc.), in order to hide some information inside before re-transmitting them. Unlike with other stealth transmission techniques, the transmission will likely be detected. However, it is expected that the transmitted signal will be confused with harmless ambient noise, and thus not identified as a communication signal.

In practice, standard audio steganography techniques are not applicable here (they are not robust to propagation in the underwater acoustic channel). Therefore, the paper focuses on an original modulation scheme, which consists to add to the initial signal a low power auxiliary signal, computed from the initial signal, whose phase and amplitude carry the information.

Finally, the possibility to reliably transmit data at a low bit rate — a few bits/sec - at a few kilometres, in a realistic underwater environment and between moving emitter and receiver, is demonstrated by a few simulations with recorded biological (clicks of sperm whales) or rain noise signals, and typical simulated doubly spread underwater acoustic channel.

Keywords: Audio watermarking, audio steganography, underwater acoustic communications, covert underwater transmission, acoustic modem.
1. INTRODUCTION

This paper addresses the application of steganography or audio watermarking [1-4] techniques to covert Underwater Acoustic Communications (UAC). More precisely, we propose to slightly modify wide band audio signals (biological noise transmitted by marine mammals, underwater works, ship noise, rain or storm noise, etc.), in order to hide binary data inside before re-transmitting them in the underwater acoustic medium. The objective of this process is to have at one's disposal a covert underwater acoustic channel, covertness meaning here that, even if the transmitted signal is detected by an enemy, this signal will likely be confused with a harmless piece of ambient noise.

In UAC, until now, covertness has often been searched by using Spread Spectrum (SS) modulations [5], particularly Direct Sequence Spread Spectrum (DSSS) with very long spreading sequence. This is only partly correct since it has been proved in [6] that, even at low signal-to-noise ratio (SNR), analysis of long transmitted DSSS signals allows to detect the transmission and to identify the spreading sequence. More recently, another covert modulation scheme, based upon OFDM, has been proposed [7]. However, even if promising, this scheme has not been proven to be more robust than DSSS to brute force detection algorithms.

In our knowledge, the first detailed description and test in UAC of a technique similar to our approach are given in a 2008 paper by Dols et al. [8], in which it is proposed to transmit simulated biological sequences of dolphin clicks, with information encoded in the delays between clicks. With respect to this technique, the interest of our approach is that, providing slight modification of the initial signal, the transmitted modified signal will appear as a perfectly realistic biological or ambient noise (not as an artificially simulated signal). Moreover, it is also guaranteed that, if the initial signal has been recorded by the emitter a few minutes before its retransmission, the transmitted signal will be perfectly compatible with the conditions of the day (no risk to transmit signal from a biological species unknown at the time and location of the transmission). Hence, if the modified signal is also transmitted at a plausible level, it will be almost impossible to identify it as carrying some information.

At first sight, the audio steganography or watermarking techniques described in the open literature are not suited to UAC, essentially because of the characteristics of the underwater acoustic channel, but also because they cannot confuse a skilled sonar operator equipped with the powerful analysis tools implemented inside modern sonar suites (sonogram with a bank of resolutions and integration times, more advanced time frequency analysis tools, etc.). For instance:

- Common techniques based upon coding of the least significant bits [1-2] are not applicable because of the expected moderate signal-to-noise ratio at the reception and, above all, since the signal is not digitized before transmission.
- Adding some auxiliary CW tones to the initial signal is not possible. Even if these tones were not immediately detected by a human listener, they can be seen on a time-frequency analysis. In a similar manner, adding short wide band pulses is no more satisfying.
- Adding delayed replicas of pieces of the initial signal (low level synthetic echoes), as proposed in [1], with information encoded in the delays is no more applicable: the delays corresponding to these echoes should be compatible with the
propagation conditions. Moreover, transmission would likely be perturbed by the actual multipaths.

Last, but not the least, it is also highly desirable that covert acoustic data link could be used for transmission between moving emitter and receiver (or at least could deal with phase drifts due to time variations of the medium and small motion of emitter and receiver near their nominal locations).

The rest of this paper addresses a general suitable modulation scheme applicable in UAC. This scheme, which leads to transmission robust to Doppler, is described in section §2, where two examples of transmitted signal (from marine mammals or rain noise) are also given. Then, section §3 is devoted to a concise description of the reception chain, with some examples of the outputs of the demodulation process on simulated data, in a noisy environment with a realistic time varying multi-paths underwater acoustic channel.

2. MODULATION SCHEME AND TRANSMISSION CHAIN

The proposed modulation scheme assumes that a previously recorded signal, \(x(t)\), is first split into a sequence of short pieces on joined non overlapping intervals with same duration \(T\) (typically a few hundred of ms). Then, the signal on each interval is processed by an “operator” \(\Psi\) which computes an auxiliary signal \(\Psi(x)(t)\) from the initial signal \(x(t)\), with approximately the same energy and bandwidth as the initial signal \(x(t)\).

Finally, a modified signal \(y(t)\), with information embedded inside it, is computed on the \(n^{th}\) interval \([nT, (n+1)T]\) by

\[
y(t) = x(t) + \alpha \cdot a_n \cdot \Psi(x)(t)
\]

where

- \(a_n\) is the embedded data (with e.g. \(a_n \pm 1\) if a BPSK scheme is used),
- \(\alpha\) is a gain which has to be taken small in order to make the transmitted signal \(y(t)\) close to the initial signal \(x(t)\).

Note that, in practice, the above operation (data insertion) is not directly performed on the initial signal \(x(t)\), but on the complex baseband signal obtained after demodulation of \(x(t)\) in a selected frequency sub-band around a nominal carrier frequency \(f_c\). It allows optimizing the frequency band with respect to transmission coveryntess and expected SNR at reception. Moreover, even it has not been done yet, it should also allow to implement modulation schemes (QPSK, QAM, etc.) with better spectral efficiency than BPSK.

Obviously, the critical point is the choice of the “operator” \(\Psi\), especially with respect to covertness and robustness to the motions of the emitter and the receiver. Here, we propose to divide each interval of duration \(T\) into \(2M\) sub-intervals, \(U_M, ..., U_1, U_1, ..., U_M\), separated by small time guards, with increasing durations, as depicted on the figure 1 below.

![Fig.1: Splitting of interval T into 2M subintervals](image-url)
Then, the auxiliary signal $\Psi(x)(t)$ is obtained from $x(t)$, by interchanging and time reverse pieces of the initial signal $x(t)$ between interval $U_k$ and $U_{-k}$, and by multiplying them by random phases terms $\exp j \beta_k$. Note that, for $\exp j \beta_k = 1$ ($k= -M, M$), the signal $\Psi(x)(t)$ is simply a time reversed copy of the initial signal $x(t)$. The product with the random phase terms $\exp j \beta_k$ (our steganographic key since the $\beta_k$ are known only by the emitter and receiver) are introduced to blur the auxiliary signal $\Psi(x)$, thus making it more difficult to detect and to rebuild by a third party.

Apart from covertness, the above “operator” $\Psi$ has been found to lead to a modulation scheme reasonably robust to the Doppler (providing proper durations for subintervals $U_k$ and time guards, and reasonable computation increase of computation load). Note also that more general schemes, with arbitrary permutations of signals from sub-interval $U_k$ to $U_l$ (with $k \neq l$) have been found not compatible with emitter and receiver motions.

The output of the above data insertion process is shown on the figure 2 below for two signals recorded at sea: a 10 seconds sequence of sperm whale clicks, and a 25 seconds record of rain noise. Spectrograms of the initial (without embedded data) and modified (with embedded data) signals are plotted figure 2. For both signals, the bits have been inserted in the frequency sub-band 1500-3000 Hz with raw bitrate 2 bits/sec (BPSK encoding with $T$ equal to 0.5 sec) and the gain $\alpha$ has been taken equal to -10 dB.

It can be seen on this figure that the spectrograms of the initial and the modified signals are almost undistinguishable. Moreover, listening of them does not show significant difference. Hence, the process used to insert the data can be here considered as covert.

![Spectrograms of initial and modified signals](image)

**Fig.2: Comparison of initial and modified signals**
3. RECEPTION CHAIN AND SIMULATION EXAMPLES

At reception, the receiver makes use of the simple three steps algorithm below:

**Step 1:** Build an approximation of the auxiliary signal $\Psi(x)(t)$ by using $\Psi(y)(t)$ (since $x(t)$ is not available; note that this approximation is correct only for small gain $\alpha$),

**Step 2:** Compute the quantity

$$z(t) \triangleq \int_{0}^{T} [\Psi(y)(u)]^{*} \cdot y(u) \cdot e^{-2i\theta u} \cdot du$$

**Step 3:** Perform synchronization (by looking for maxima $|z(t)|$) and recover the transmitted data from the phases of $z(t)$ at the times of its maxima.

In practice this algorithm is applied to the complex base-band signal obtained after demodulation of the received acoustic signal at the nominal carrier frequency $f_c$. For moving emitter or receiver, i.e., for non-zero Doppler shift, the nominal $f_c$ and the actual carrier frequency $f_c'$ are not equal. Hence, an additional term $e^{-2i\bar{\theta}u}$ is used to compensate the residual of carrier $f_c'$-$f_c$. Note that the parameter $\bar{\theta}$ is determined, at every time $t$, by maximizing $|z(t)|$ on a grid $\{\theta_p; 1 \leq p \leq P\}$ covering the plausible Doppler shifts.

It is easy to check that, when applied to the modified transmitted signal $y(t)$ with the proper operator $\Psi$ (including the steganographic key $\{\beta_k\}$), the quantity $|z(t)|$ exhibits periodic peaks, at period $T$ and with phases and amplitudes proportional to the transmitted data $a_n$. When applied to the acoustic signal received after propagation in the underwater medium, these periodic peaks are no more Dirac pulse, but are replaced by the autocorrelation of the channel impulse response $h(t)$.

To illustrate the above points, we have simulated the signal received after propagation along the doubly spread simulated underwater acoustic channel shown plotted on the figure 3 below.

![Bandlimited channel impulse response](image)

**Fig.3:** Simulated underwater acoustic channel
This channel has been simulated using the technique described in [9], with main parameters as follows:

- carrier frequency: 2250 Hz,
- emitter/receiver range: 5000 m
- water depth 100 m, emitter an receiver depth: 5 and 60 meters
- doppler spread: 0.5 Hz

It corresponds to a typical moderately difficult shallow water acoustic channel.

The output of the step 2 of the demodulation process – i.e. the quantity $|z(t)|$ - is plotted figure 4 for the two signals considered in §3, after propagation in the simulated channel of figure 3 and addition of a Gaussian noise, with signal-to-noise ratio (SNR) equal to 0 dB (ratio of the power of the whole transmitted signal $\Psi(x)(t)$ to the power of the noise). The expected peaks, at rate $T$, can be observed on the two plots of figure 4, thought with a poorer contrast for the whale clicks than for the noise of rain.

In order to simulate a non-zero Doppler – here a radial speed equal to 1.7 m/s – the transmitted signal $y(t)$ has been resampled (resampling rate $1 + \frac{v}{c_{el}}$, with $c_{el}$ equal to 1500 m/s) before convolution with the channel response.

![Figure 4: Output of the demodulation process (step 2)](image)

The estimated symbols - output of the step 3 of the demodulation process - are plotted figure 5, after synchronisation and phase compensation (using a known symbol transmitted at the beginning).
It can be seen on this figure that, even with low SNR and gain $\alpha$, the symbols are properly recovered by the demodulation algorithm. It is also worth noticing that, for both transmitted signals, the 16 or 50 transmitted bits were all recovered without error.

4. CONCLUSIONS

A novel approach has been proposed for covert underwater acoustic communication. It is based upon re-transmission of previously recorded pieces of ambient noise, after a slight modification in order to hide some information inside them.

The results of a few simulations indicate that the covertness of this approach is correct (modification of the signal difficult to detect by listening or time/frequency analysis). This approach allows to transmit information with low bitrate – a few bits per second – providing a signal-to-noise ratio about 0 dB at reception. It is also reasonably robust with respects to emitter or receiver motions (Doppler),

Future works should be devoted to improvement of the modulation and demodulation algorithms, to experimental validation by at sea experiments and finally to thorough analysis of the proper manner to select the used pieces of ambient noise (both for covertness and transmission efficiency).
REFERENCES


THE DESIGN OF WIDE BAND TRANSDUCERS FOR UNDERWATER ACOUSTIC COMMUNICATION

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Abstract: To implement effective underwater communication, it is essential to design wide band transducers with compact structures. Tube transducers have been used as hydrophones for underwater acoustic measurements. By combining different sizes of PTZ elements, this type of transducers can be applied in wide frequency range with moderate transmitting response and sensitivity. In this paper, the working principles of transducers are introduced, and the transducers in different frequency ranges are manufactured. The measurement results suggest that this type of transducers can be designed to cover the frequency range from 10 kHz to 80 kHz.

Keywords: underwater acoustics communication, hydrophone, tube transducers, wide band
1. Introduction

Piezoelectric ceramic tubes have been used in designing different types of ring transducers, due to its stable performance, omni-directivity in horizontal plane and ease in disposition\cite{1,2}. Up to date, the ring transducers have been brought into wide applications. Generally, to get high source level, a ring transducer should be driven at the frequency near its radial vibration mode. However, the bandwidth of the radial vibration mode is relatively limited. To use ring transducers as wide band sources, besides the radial vibration, other vibration modes should play their roles as well. Resonance of the liquid cavity of a ring transducer can be coupled with radial vibration mode. However, the related resonance will only appear obviously in the frequency range lower than 10 kHz\cite{3}. Higher order radial vibrations of tubes can be excited to transmit acoustic waves in the high frequency range, whereas the horizontal directivity of the transducers will become uneven\cite{4}.

In this paper, the flexural vibrations along the height of tubes are used to expand the bandwidth of transducers. The simulation based on the finite element method (FEM) reveals that by selecting suitable dimension of a tube, its radial and flexural vibration will appear subsequently in the expected band. In the frequency range from 10 kHz to 80 kHz, the wide band transducers can be designed using the same principle. Experimental results confirm that this type of transducers can be used as wide band transceivers as well.

2. The principle of wide band transducer

For a transducer driven on its radial vibration mode, its bandwidth can be expressed through the mechanical quality factor \( Q_m \):

\[
Q_m = f_r / \Delta f = M_e \omega / R
\]

where \( f_r \) is the resonant frequency, \( \Delta f \) is the band width when conduct response decreases 3dB, \( \omega \) is the angular frequency, \( R \) is the sum of radiation resistance, mechanical wear resistance and dielectric loss resistance of the transducer, \( M_e \) is the equivalent mass of transducer.

In order to expand the band width of a ring transducer, the value of \( R \) should be increased and the equivalent mass \( M_e \) should be reduced. Generally, it is difficult to acquire \( Q_m < 3 \) for ring transducers in single vibration mode, so the flexural vibration should be brought and set in the range close to the radial vibration. Through the coupling of the two vibration modes, a ring transducer can be used in the relatively wide band.

It is found that by adjusting the ratio of thickness and height of a tube, the resonant frequency in flexural vibration will change accordingly. The parameters of transducers can be calculated with classical theoretical methods. The FEM provide an intuitive tool to set working bandwidth by analyzing each vibration mode. For a PZT-4 tube with an inner and outer diameter as 32 mm and 36 mm, the radio vibration frequency is calculated as 30.4 kHz,
and its flexural vibration is 34.3 kHz for a height of 22 mm. From figure 1, it can be found
that with increase of the height of a tube, the frequency of flexural vibration will reduce,
while the radial vibration will keep at almost the same frequency.

Figure 1: (a) Radial transducer resonant mode and (b) high-order bending mode of ring transducer

3 The design of ring transducer

Two PZT tubes with an inner and outer diameter of 32 mm and 36 mm, the height of 22
mm are connected in parallel to make a wide band transducer. The admittance, transmitting
voltage and receiving sensitivity are simulated through FEM software and shown in figure 2.

Figure 2: (a) Admittance curve, (b) transmitting voltage response, and (c) sensitivity response of a ring
transducer
The simulation reveals that the resonant frequencies of radial and flexural vibration are 27 kHz and 32 kHz, respectively, and the fluctuation of transmitting voltage response is about ± 2.5 dB in the range of 21.5 kHz to 40 kHz, and about ± 2.5 dB for receiving sensitivity in the range from 20 kHz to 36.5 kHz.

4 The performance of ring transducer

By selecting different sizes of PZT tubes, a series of wide band transducers are designed and manufactured in the frequency range from 10 kHz to 80 kHz. Based on the FEM simulation, three types of transducers are designed and manufactured, which cover three bands: 10-20 kHz, 20-40 kHz, 40-80 kHz, respectively. The transmitting voltage responses and receiving sensitivities of three types of transducers are measured and illustrated in the figure 3, 4 and 5.

Figure 3: Transmitting voltage response and receiving sensitivity of a 10-20 kHz transducer

Figure 4: Transmitting voltage response and receiving sensitivity of a 20-40 kHz transducer

Figure 5: Transmitting voltage response and receiving sensitivity of a 40-80 kHz transducer
Measurement results confirm that the ring transducers can be developed as wide band transceivers by coupling radial and flexural vibration in a desired frequency range. With the serialized three transducers covering the range 10-80 kHz, the transmitting voltage response can be kept in the range from 130 dB to 145 dB, while the receiving sensitivity will fluctuate in the range from -190 dB to -210 dB.

5 Conclusion

The radial vibration of PZT ceramic tubes is the primary mode in ring transducers. By exciting flexural vibrations of tubes and coupling diverse modes, the wide band transducers can be achieved in the frequency range from 10 kHz to 80 kHz.

By adjusting the ratio of thickness and height of a tube, the resonant frequencies for radical and flexural vibration can be set at a suitable interval, which will determine both the working bandwidth and transmitting response of the transducer.

Both the theoretical simulation and experimental results confirm that ring transducers with flexural vibration mode have relatively flat transmitting response and sensitivity, which make them suitable to be used as transceivers.

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MEASUREMENT AND MODELING OF FADING IN ULTRASONIC UNDERWATER CHANNELS

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\textit{Abstract:} This paper reports measurements for ultrasonic underwater acoustic communication (UAC) channels in Mediterranean shallow waters performed by the company SAES and the research group Ut\textsuperscript{2} of the University of Málaga. Statistical fit to the measured data is also carried out. The measurements have been conducted when the transmitter and the receiver are spaced 50, 100 and 200 m, approximately. Both transducers (B\&K 8105 and RESON TC4032) have been placed at depths 3, 6 and 9 m from anchored boats. Faded channel sounding signals have been recorded at frequencies 32, 64 and 128 kHz. After preprocessing the measured data, accurate statistical fit to the recently proposed $\kappa\mu$ shadowed fading model is obtained; this model includes, among others, the Rayleigh, Ricean and Nakagami-$m$ fading.

\textit{Keywords:} underwater acoustic communications, measurements, fading, $\kappa\mu$ shadowed statistical model
1. INTRODUCTION

A crucial aspect for underwater acoustic communication (UAC) is the statistical characterization of the communication channel. High-frequency UAC for short-range applications is very challenging; the promise of small-size transducers has the drawback of fast time-varying channels. Shallow waters UAC channels experience significant fading even when the transmitter and the receiver are not intentionally moving relative to each other. A number of researchers have measured and modeled fading UAC channels in the audio band [1]-[2]; however, fading of ultrasonic UAC channels has been barely investigated. Such channels appear in UAC systems which employs small-size transducers (and consequently with high resonance frequency), e.g. diver-to-diver communications or networks with small-size sensors.

First, measurements for ultrasonic UAC channels are reported in Mediterranean shallow waters, performed by the company SAES (www.electronica-submarina.com) and the research group Ut² of the University of Málaga (www.ut2.uma.es). Secondly, statistical fit to the measured data is performed. The measurements have been conducted when the transmitter and the received are spaced 50, 100 and 200 m, approximately. Both transducers (B&K 8105 and RESON TC4032) have been placed at depths 3, 6 and 9 m from anchored boats. Faded channel sounding signals have been recorded at frequencies 32, 64 and 128 kHz. After preprocessing the measured data, statistical fit to the recently proposed $\kappa-\mu$ shadowed fading model is performed [3]; this model includes, among others, the Rayleigh, Ricean and Nakagami-$m$ fading.

The remainder of this paper is organized as follows. In Section 2 the measurements of ultrasonic UAC channels are described. The statistical fit of the observed fading to the $\kappa-\mu$ shadowed model is presented in Section 3. Finally, some conclusions on our research results are provided in Section 4.

2. MEASUREMENT OF ULTRASONIC UAC FADEING CHANNELS

This work is part of a measurement campaign named underwater communication experiments (UCEX). UCEX is a multiyear measurement project of the company SAES and the University of Málaga.

The general scenario considered in this paper is the following: Mediterranean shallow waters (depths from 14 to 30 m), sandy seabed, separations of 50, 100 and 200 m between the transmitter and the receiver; and the projector and the hydrophone suspended from 7 m length anchored boats, both at the same depth (3, 6 or 9 m). Since the main goal of this work is the analysis of the fading statistics, the probe signals employed are sinusoidal of frequencies 32, 64 and 128 kHz. All the measurements in this work were obtained on November 13 (2013) in La Algameca Chica, Cartagena (Spain). The specific data of the different channels considered along this work are provided in Table 1, including a short channel code for future reference.
The equipment developed for the UCEX measurement campaign uses the projector Bruel&Kjaer 8105 and the hydrophone Reson TC4032, performing accurate noise and channel measurements in the 32-128 kHz ultrasonic band. For each record the sample rate is 1 MHz and the total time duration is 60 s. The records obtained with the UCEX measurement system are digitally post-processed as follows. First, a 400 Hz bandwidth bandpass filter centered at the frequency of the transmitted sinusoidal signal is applied. This bandwidth must be larger than the observed Doppler bandwidth in the experiments and, simultaneously, small enough for cleaning the receiver signal of low frequency and impulsive noise. Then, a simple envelope detector is employed to obtain the normalized channel power gain \( \alpha^2 \) (i.e. \( E[\alpha^2]=1 \)), where \( \alpha \) represents the channel envelope. The cumulative distribution function (CDF) of the normalized power gain \( \alpha^2 \) is defined as

\[
F_{\alpha^2}(x) = \Pr[\alpha^2 \leq x].
\]  

(1)

Fading of the ultrasonic UAC channels shown in Table 1 is fully characterized by the CDF of the power gain. From Fig. 2 to Fig. 4, the experimental CDFs of the channels given in Table 1 are presented. From the point of view of performance of fading channels it is convenient to represent the CDF of the normalized power gain in a log-log plot; the reason is that the outage probability, average bit-error rate or outage capacity of the channel is primarily determined by the statistics of the fading events. The statistical shape of the measured channels shown in Fig. 2 (distance 50 m) exhibit moderate variations with the frequency; however, channels in Fig. 3 (distance 100 m) are nearly independent of the frequency. Comparing Fig. 2 and Fig. 3 one concludes that the distance between the transmitter and the receiver clearly impact on the shape of the CDF. Fig. 4 presents the measured normalized power gain for the channels with a separation of 200 m between the transmitter and the receiver; it is clearly shown that the depth of both transducers has a great influence on the fading statistics (channel C3-64 versus C9-64).

### Table 1: Parameters and conditions of the ultrasonic UAC channels.

<table>
<thead>
<tr>
<th>Channel code</th>
<th>Frequency [kHz]</th>
<th>Transducers’ depth [m]</th>
<th>Transducers’ separation [m]</th>
<th>Average sea depth [m]</th>
<th>WMO sea state</th>
</tr>
</thead>
<tbody>
<tr>
<td>A6-32</td>
<td>32</td>
<td>6</td>
<td>50</td>
<td>16</td>
<td>2</td>
</tr>
<tr>
<td>A6-64</td>
<td>64</td>
<td>6</td>
<td>50</td>
<td>16</td>
<td>2</td>
</tr>
<tr>
<td>A6-128</td>
<td>128</td>
<td>6</td>
<td>50</td>
<td>16</td>
<td>2</td>
</tr>
<tr>
<td>B6-32</td>
<td>32</td>
<td>6</td>
<td>100</td>
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<td>2</td>
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<tr>
<td>B6-64</td>
<td>64</td>
<td>6</td>
<td>100</td>
<td>20</td>
<td>2</td>
</tr>
<tr>
<td>B6-128</td>
<td>128</td>
<td>6</td>
<td>100</td>
<td>20</td>
<td>2</td>
</tr>
<tr>
<td>C3-64</td>
<td>64</td>
<td>3</td>
<td>200</td>
<td>25</td>
<td>2</td>
</tr>
<tr>
<td>C9-32</td>
<td>32</td>
<td>9</td>
<td>200</td>
<td>25</td>
<td>2</td>
</tr>
<tr>
<td>C9-64</td>
<td>64</td>
<td>9</td>
<td>200</td>
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<td>C9-128</td>
<td>128</td>
<td>9</td>
<td>200</td>
<td>25</td>
<td>2</td>
</tr>
</tbody>
</table>
Fig. 2: CDFs of the ultrasonic UAC channels A6-32, A6-64 and A6-128.

Fig. 3: CDFs of the ultrasonic UAC channels B6-32, B6-64 and B6-128.
3. MODELING THE ULTRASONIC UAC CHANNELS

The measured ultrasonic UAC channels are modeled in this Section by the \( \kappa-\mu \) shadowed fading distribution; this model has recently proposed in [3]. Since the \( \kappa-\mu \) shadowed distribution includes the Rayleigh, Rician, Rician shadowed and \( \kappa-\mu \) distribution, it provides a very flexible model to fit the experimental data. The CDF of the \( \kappa-\mu \) shadowed model for the normalized channel power gain is given by [3]

\[
\hat{F}_{c;\kappa,\mu,m}(x) = \frac{\mu^{\kappa} m^m (1+\kappa)^\mu}{\Gamma(\mu)(\mu\kappa+m)^m} x^{m} \\
\times \Phi_2 \left( \mu-m, m; \mu+1; -\mu(1+\kappa)x, -\mu(1+\kappa) \frac{mx}{\mu\kappa+m} \right),
\]

where \( \mu \) represents the effective number of clusters, \( \kappa \) is the ratio between the average power of the dominant components and the scattered components, \( m \) is the parameter associated to the power fluctuation of the dominant components and \( \Phi_2 \) the bivariate confluent hypergeometric function [4]. The fit procedure of the CDF in eq. 2 to the experimental data consists on numerically solving the following optimization problem

\[
\min_{\kappa,\mu,m} \left( \max_{x} \left| \log_{10} \frac{\hat{F}_{c;\kappa,\mu,m}(x)}{F_{c;\kappa,\mu,m}(x)} \right| \right).
\]

The fit procedure expressed in eq. 3 consists on minimizing the maximum relative deviation between the experimental CDF of the power gain \( F \) and the \( \kappa-\mu \) shadowed CDF \( \hat{F} \) and can be carried out by standard numerical methods. The result of the \( \kappa-\mu \)
shadowed modeling is given in Table 2, where the maximum deviation achieved in the fit procedure is indicated.

<table>
<thead>
<tr>
<th>Channel code</th>
<th>$\kappa$</th>
<th>$\mu$</th>
<th>$m$</th>
<th>$\max_{x} \log_{10} \frac{F_{\hat{\kappa},\mu}(x)}{F_{\kappa}(x)}$</th>
</tr>
</thead>
<tbody>
<tr>
<td>A6-32</td>
<td>2.33</td>
<td>0.92</td>
<td>1.86</td>
<td>0.029</td>
</tr>
<tr>
<td>A6-64</td>
<td>2.90</td>
<td>1.00</td>
<td>3.10</td>
<td>0.068</td>
</tr>
<tr>
<td>A6-128</td>
<td>9.56</td>
<td>1.27</td>
<td>1.67</td>
<td>0.114</td>
</tr>
<tr>
<td>B6-32</td>
<td>3.03</td>
<td>0.91</td>
<td>2.15</td>
<td>0.056</td>
</tr>
<tr>
<td>B6-64</td>
<td>1.89</td>
<td>0.94</td>
<td>1.32</td>
<td>0.049</td>
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<tr>
<td>B6-128</td>
<td>1.99</td>
<td>1.01</td>
<td>1.86</td>
<td>0.022</td>
</tr>
<tr>
<td>C3-64</td>
<td>7.71</td>
<td>0.90</td>
<td>18.01</td>
<td>0.205</td>
</tr>
<tr>
<td>C9-32</td>
<td>4.06</td>
<td>1.13</td>
<td>2.45</td>
<td>0.026</td>
</tr>
<tr>
<td>C9-64</td>
<td>0.03</td>
<td>1.02</td>
<td>6.32</td>
<td>0.053</td>
</tr>
<tr>
<td>C9-128</td>
<td>1.56</td>
<td>1.04</td>
<td>2.39</td>
<td>0.040</td>
</tr>
</tbody>
</table>

*Table 2: Modeling the ultrasonic UAC channels by the $\kappa$-$\mu$ shadowed distribution.*

4. CONCLUSIONS

This work presents both measurement and modeling of ultrasonic UAC channels performed by SAES (www.electronica-submarina.com) and the research group Ut$^2$ of the University of Málaga (www.ut2.uma.es). Our main conclusions are that fading in ultrasonic UAC is very significant and that the $\kappa$-$\mu$ shadowed fading model is quite appropriate for its statistical characterization.

ACKNOWLEDGEMENTS

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REFERENCES

JOINT PILOT BASED CHANNEL ESTIMATION WITH SELECTED-MAPPING TO REDUCE PAPR IN UNDERWATER ACOUSTIC MIMO-OFDM SYSTEM WITHOUT SIDE INFORMATION

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Abstract: In this paper, a peak-to-average-power ratio reduction scheme that requires no side information in underwater acoustic MIMO-OFDM system is proposed. The key idea of the scheme is that different phase sequences is represented by different comb pilot sequences, and it uses channel estimation scheme to distinguish the number of phase sequences while completing the channel estimation at the receiver. Therefore, the proposed scheme does not need to reserve bits for transmitting side information, so that the data rate can be increased. Simulation results show that the PAPR reduction performance was not reduced when comparing with the traditional selected mapping and the BER performance is approximately the same as those selected mapping scheme with perfect SI. Experiment carried in the tank also demonstrates the proposed scheme can differentiate phase sequences, and significantly enhance the quality of the underwater acoustic MIMO-OFDM communication system.

Keywords: Underwater acoustic communication, MIMO-OFDM, channel estimation, Selected Mapping, Side Information
MIMO-OFDM technique is regarded as the most possible solution that can improve the spectral efficiency without enhancing the transmitting power or expanding the communication bandwidth for high speed underwater acoustic communication [1]. However, MIMO-OFDM system suffers several drawbacks such as high peak-to-average power ratio (PAPR). High PAPR requires large liner region of the amplifier, or it will lead to distortion. Therefore, many PAPR reduction schemes have been proposed [2].

Among them, the Selected Mapping (SLM) scheme, which is proposed by R. W. Bauml in 1996, is a widely used scheme to reduce the PAPR without signal distortion [3]. In SLM scheme, the input data block is multiplied with \( M \) different phase sequences which produce a modified data block. Then the OFDM signal with the lowest PAPR is selected for transmission after the IFFT of \( M \) independent sequences was taken. The index of the selected phase sequence should be transmitted as side information (SI) when the data symbols are sent. In order to let the receiver be able to recover the original data block, the SI should be protected by channel coding or other special coding methods, which can result in system complexity increasing, data transmitting rate losing and communication quality decreasing.

A selective time-domain filtering technique is proposed in [4], the receiver can recover the data only by using conventional channel estimation and demodulation techniques without SI. It requires more pilot symbols which can decrease the band efficiency. Hyun-Seung Joo et al. proposed the blind SLM PAPR reduction scheme using cyclic shift in [5]. It uses the cyclic shift and linear combining instead of multiplying phase sequences, resulting in system complexity decreasing. However, this method cannot efficiently increase the BER performance of the system. A low-complexity PAPR Reduction scheme without side information is proposed by Seung-Sik Eom in [6]. In this scheme, circularly shifted phase sequences and phase rotation are applied to create OFDM symbol with low PAPR. A receiver that demodulates the transmitted source symbols without SI in the ML fashion is also designed. Although this method has less computation and no side information transmission, it requires channel state information as prior information. It is also difficult achieving channel estimation and equalization because of no reserved pilot sub-carriers. Moreover, a series of timing and synchronization issues may be caused, so this algorithm is not suitable for underwater acoustic communication.

In this paper, we proposed a modified SLM scheme that joint pilot based channel estimation with SLM to reduce the PAPR in underwater acoustic MIMO-OFDM system. The proposed scheme requires no side information. The key idea is that different phase sequences is represented by different comb pilot sequences, and it uses channel estimation scheme to distinguish the number of phase sequences while completing the channel estimation at the receiver. Therefore, there is no need to reserve bits for transmitting side information, so that the data rate can be increased. The modified scheme has almost no degradation of PAPR reduction performance when compared with the traditional SLM method, and the BER performance is approximately the same as those selected mapping scheme with perfect SI.
**SYSTEM MODEL**

The framework of the modified PAPR reduction scheme in MIMO-OFDM system is shown in figure 1. The system adds different comb pilot sequences which represent different phase sequences, and selects the OFDM signal with minimum PAPR to transmit. The main part of the receiver is the phase sequence detector, which uses the channel estimation scheme to distinguish the number of comb pilot sequences while completing the channel estimation. According to the correspondence between the comb pilot sequences and the phase sequence, which phase sequence was selected at the transmitter can be known.

![Figure 1: Framework of the modified Selective Mapping PAPR Reduction scheme in MIMO-OFDM system. (a) The transmitter (b) The receiver](image)

**PROPOSED SLM SCHEME**

Take one symbol as an example to explain the principle of the proposed SLM scheme. The number of sub-carriers is $N$, the number of pilot tones is $N_p$ in MIMO-OFDM system. Denote $Po = \{i_0, i_1, \cdots, i_{N_p-1}\}$ as the location of pilot tones and $Po^C$ as the complementary set of $Po$ in $R = [0, 1, \cdots, N - 1]$, represent the data symbols location. The input data block $X_D = [X_D(0), X_D(1), \cdots, X_D(N-1)]$ is multiplied with $M$ different phase sequences $P^m = [P^m(0), P^m(1), \cdots, P^m(N-1)]^T$, $0 \leq m \leq M$, which produce a modified data block $X_D^m = P^mX_D$. Obviously, $X_D^m \equiv 0$ when $k \notin Po^C$.

Define a set of different pilot sequences $X_P^m = [X_P^m(0), X_P^m(1), \cdots, X_P^m(N-1)]$ corresponding to different phase sequences, $X_P^m(k) \equiv 0$ when $k \notin Po$. Therefore, we have
\[ X^m(k) = X^m_D(k) + X^m_P(k) = \begin{cases} X^m_D(k), & k \in Po^C \\ X^m_P(k), & k \in Po \end{cases} \]  \tag{1}

IFFT of \( M \) independence sequences \( X^m(k) \) are taken to produce the sequences \( x^0, x^1, \ldots, x^{M-1} \), among which the one \( x \) with the lowest PAPR is selected for transmission.

After taking the fast Fourier transform (FFT) operation at the receiver, the received signal in frequency domain can be given as

\[ Y(k) = X(k)H(k) + W(k) \]  \tag{2}

Where \( H(k) \) denotes the frequency response of the underwater acoustic channel, and \( W(k) \) represents the white Gaussian noise in frequency domain. Denote \( \hat{H}^m(k) \) as the estimation of \( H(k) \), since the value and location of all the pilot tones is known, the sampling value of the underwater acoustic channel frequency response can be achieved by

\[ \hat{H}_P^m(k) = \frac{Y_P}{X_P^m} \]  \tag{2}

Where \( Y_P \) is the received data on the pilot tones in frequency domain. Then, \( \hat{H}^m(k) \) can be obtained by linear interpolation.

IFFT of \( M \) frequency response \( \hat{H}^m(k) \) are taken to produce the channel impulse response \( \hat{h}^m \), among which the value of \( m \) matches \( \hat{h} = \max_{0 \leq m \leq M-1} \left\{ \| \hat{h}^m \|_o \right\} \) is the number of selected phase sequence at the transmitter. The corresponding phase sequence \( P^m \) is the answer we are looking for.

**SIMULATION AND EXPERIMENTAL RESULTS**

Simulations are carried out to validate the feasibility of the proposed algorithm. Four sparse channels generated from channel simulation software are used in simulation to evaluate the performance. The impulse response of the simulation channels are shown in the Fig.2, while the main parameters of MIMO-OFDM system are listed in Table 1. The
information is divided and coded according to the information content of an OFDM symbol. The number of phase sequences is $M=4$.

<table>
<thead>
<tr>
<th>Parameters</th>
<th>Value</th>
<th>Parameters</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>FFT length</td>
<td>8192</td>
<td>Band Range</td>
<td>6–12kHz</td>
</tr>
<tr>
<td>Sub-carrier Number</td>
<td>1025</td>
<td>Sub-carrier spacing</td>
<td>5.86Hz</td>
</tr>
<tr>
<td>Transmitter/Receiver Number</td>
<td>$2 \times 2$</td>
<td>Code method</td>
<td>STBC &amp; Convention Code</td>
</tr>
<tr>
<td>OFDM length</td>
<td>171ms</td>
<td>Cyclic Prefix Length</td>
<td>43ms</td>
</tr>
<tr>
<td>Sample Rate</td>
<td>48kHz</td>
<td>Code Rate</td>
<td>1/2</td>
</tr>
</tbody>
</table>

Table 1 Parameters of MIMO-OFDM system

![Table 1 Parameters of MIMO-OFDM system](image)

Fig. 2 The impulse response of the simulation channels

It can be seen in Fig.3 that the PAPR reduction performance of the proposed scheme is not degraded when comparing with the traditional selected mapping and the BER performance is better. It also can be seen that the BER performance is approximately the same as those selected mapping scheme with perfect SI.

![Fig. 3: Comparisons of the performance between the traditional and the proposed scheme.](image)

(a) PAPR reduction performance (b) BER performance

For the purpose of validating the feasibility of improved SLM algorithm, the experiment is carried out in the underwater tank of acoustic lab, Harbin Engineering University, November 2013. The real impulse response of the tank channel is shown in Fig.5.
Fig.5 Real impulse response of tank channel

Fig.6 shows the experimental result of the first 40 OFDM symbols in the transmitted signal. Fig.6 (a) is the original version of side information which is produced at the transmitter. The output, which is autonomous recognized by phase sequence detector at the receiver, is shown in Fig.6 (b). The gradation value shown in Fig.6 is inversely proportional to the autocorrelation value of the estimated channel impulse response.

It has been verified that the proposed scheme can differentiate among phase sequences with error rate below 0.7% from the comparison between Fig.6 (a) and Fig.6 (b), even channel state information and side information are unknown. Therefore, the whole symbol decoding errors caused by side information mistranslated can be avoided, and the communication quality can be guaranteed by combining the coding method with the high code rate.

CONCLUSION

A peak-to-average-power ratio reduction scheme that requires no side information is proposed in underwater acoustic MIMO-OFDM system. A set of different pilot sequences is defined corresponding to different phase sequences in the proposed scheme. Channel estimation scheme is used to distinguish the number of phase sequences while completing the channel estimation at the receiver. There is no need for the proposed scheme to reserve bits for transmitting side information, so that the data rate is increased. The simulation and experiment results show that the performance of the proposed scheme is approximately the same as those selected mapping scheme with perfect SI and significantly enhance the quality of the underwater acoustic MIMO-OFDM communication system.
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REFERENCES

ULTRASONIC DIVERSITY OFDM TRANSCEIVER ARCHITECTURE WITH IMPULSIVE NOISE CANCELLING FOR SHALLOW SEA COMMUNICATION

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Abstract: In order to support shallow sea underwater communication to explore marine natural resources using remote robotic control or to enable rapid information exchange between divers and so on, a robust Digital Communication method under the multiple delayed refraction wave circumstance is necessary. We propose OFDM Ultrasonic communication system with Diversity receiver. It utilizes 20-28 (KHz) ultrasonic channel and Subcarrier Spacing of 46.875 (Hz), 161-subcarriers OFDM modulation. Living creatures in shallow sea generate Impulsive Noise so called Shrimp Noise. Then Our OFDM diversity receiver has Time and Frequency Domain Impulsive noise Canceller with Maximum ratio combiner. The paper shows the proposed Diversity OFDM Transceivers architecture and Experimental results taken at a fishing port in Okinawa Japan, which has shown QPSK communication more than 50m distance shallow sea. In addition, an Inter-Carrier Interference Canceller is incorporated, and experiments with moving receivers at 0.6 (m/s) and 0.9 (m/s) are conducted as well.

Keywords: OFDM, Impulsive Noise, Inter-Carrier Interference, Maximum Ratio Combiner
1. INTRODUCTION

While underwater communication brings great benefits, the underwater transmission poses many challenges, especially for a high speed communication system. First, though acoustic signal rather than high frequency radio signal is used for underwater communication, the signal strength still degrades quickly. Second, the speed of acoustic signal in water environment is about 1500 m/s, so the signal will suffer very a long delay and high Doppler spread that causes ICI. Finally, the presence of impulsive noise also degrades reception signal.

To solve those four challenges, we propose an ultrasonic OFDM system consists of four diverse receivers [1], 2D-DFT channel estimation, ICI cancelling [2-4], and combining time and frequency impulsive cancelling [5-6]. Fig. 1 shows the whole proposal system.

The rest of this paper is organized as follow. Section 1 describes the proposal system architecture. Experimental results are shown in Section 2. Finally, conclusion is presented in Section 3.

![Proposal Transceiver Architecture](image)

**Fig. 1: Proposal Transceiver Architecture**

<table>
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<th>Parameters</th>
<th>Mode</th>
<th>2</th>
<th>3</th>
</tr>
</thead>
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<td>TX-RX Elements</td>
<td></td>
<td>1 TX and 4 RX Transducer</td>
<td></td>
</tr>
<tr>
<td>Sampling Frequency</td>
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<td></td>
</tr>
<tr>
<td>TX Center Frequency</td>
<td></td>
<td>24000 Hz</td>
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</tr>
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<td>Band Width</td>
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<td>8000 Hz</td>
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<td>2048</td>
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<td></td>
<td>10.667 ms</td>
<td>21.333 ms</td>
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<td>Sub Carrier Spacing</td>
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</tr>
<tr>
<td>Number of Sub Carrier</td>
<td></td>
<td>81</td>
<td>161</td>
</tr>
</tbody>
</table>

*Table 1: System Parameters*
2. THE PROPOSAL SYSTEM DESCRIPTION

2.1. Pilot Structure and 2D-DFT Channel Estimation

Fig. 2 shows the pilot structure. Channel is measured by scattered pilots in both time and frequency domain. As shown in Fig. 2, with any given sub-carrier, channel is measured by pilots spacing 3 OFDM symbol in time domain. Since channel transfer function (H) at pilot positions is measured, a 2D-DFT interpolation is employed to interpolate the whole channel transfer function at all positions. After the 2D-DFT process, a 2D Doppler-Delay profile as Fig 3 including a real Doppler-Profile and three aliases. Those aliases have to be removed before converting back the Doppler-Profile to channel transfer function.

2.2. Inter-Carrier Interference Cancellation

According to [1-3], the ICI equation can be written as follow

$$Y(k) = X(k) \cdot H_{k,k} + \sum_{l=0, l \neq k}^{N-1} X(l) \cdot H_{k,l} + W(k)$$  \hspace{1cm} (1)

X(k) and Y(k) denotes transmitted and received signal, respectively and l, k are sub-carrier indexes. N is number of sub-carriers and W(k) is random noise. \(H(k, l)\) denotes the channel transfer function from \(l^{th}\) to \(k^{th}\) sub-carrier, and is calculated as (2).

$$H_{k,l} = \frac{1}{N} \sum_{n=0}^{N-1} \sum_{m=0}^{N-1} h(m, n) \cdot e^{-j \frac{2\pi lm}{N}} \cdot e^{-j \frac{2\pi k}{N}(k-l)}$$  \hspace{1cm} (2)

Here, \(h(m, n)\) is channel impulse response, with \(m\) and \(n\) are delay path and time index, respectively. The main idea in [1-3] is to linearly approximate the time varying channel \(h(m, n)\) within one OFDM symbol, then \(H(k, l)\) can be determined. Then, ICI can be removed from received subcarrier. In more detail, please refer to [2-4].
2.3. Frequency Diversity MRC

Due to multipath effect, signal of a given sub-carrier can be constructed or destructed at different places. So, a four diverse receivers system based on MRC [1] is proposed to combine signal from four branches after equalization. Following [1], to compute combing coefficient, noise is taken into account as follow

\[
\text{MRC}(k) = \frac{EQ(k)_1 \cdot \frac{|H_{kk1}|^2}{\sigma^2_{n1}} + EQ(k)_2 \cdot \frac{|H_{kk2}|^2}{\sigma^2_{n2}} + EQ(k)_3 \cdot \frac{|H_{kk3}|^2}{\sigma^2_{n3}} + EQ(k)_2 \cdot \frac{|H_{kk4}|^2}{\sigma^2_{n4}}}{\frac{|H_{kk1}|^2}{\sigma^2_{n1}} + \frac{|H_{kk2}|^2}{\sigma^2_{n2}} + \frac{|H_{kk3}|^2}{\sigma^2_{n3}} + \frac{|H_{kk4}|^2}{\sigma^2_{n4}}}
\]

(3)

Here \(\sigma^2_{n1}, \sigma^2_{n2}, \sigma^2_{n3},\) and \(\sigma^2_{n4}\) are estimated average noise power at branch 1, branch 2, branch 3, and branch 4, respectively.

2.4. Time and Frequency Impulsive Noise Cancelling

Because the presence of impulsive noise so-called shrimp noise, time and frequency noise cancelling (time-IMP, freq.-IMP) methods [5-6] are employed in the proposal system. Time-IMP detects impulsive noise samples in time domain, then clips the impulsive noise samples as follow

\[
y(n) = \begin{cases} 
  r(n) & \text{if } |r(n)|^2 \leq 5 \cdot P_{avg} \\
  0 & \text{if } |r(n)| > 5 \cdot P_{avg}
\end{cases}
\]

(4)

Here \(r(n)\) is \(n^{th}\) time sample of received signal after pre-amplifier, \(P_{avg}\) is the average power of an OFDM symbol. In addition, a frequency impulsive noise cancellation [6] is applied. For more detail about the freq.-IMP, please refer to [6].

3. EXPERIMENTAL RESULTS

The experimental parameters are shown in Table 2 and Fig. 4. Fig. 5-8 compares bit error rate among 1 single receiver (1BR), 2 diverse receivers (2BR) and four diverse receivers (4BR). The black square marked line means T-IMP and Freq-IMP are OFF, and no ICI cancelling. The blue triangle marked line means T-IMP and Freq-IMP are OFF, and ICI cancelling is applied. Finally, the red line is T-IMP and Freq-IMP are ON, and ICI cancelling is applied. In addition, Fig. 5 and 6 show Mode 3 when receivers are no moving and moving at 0.6 (m/s), respectively. Similarly, Fig. 7 and 8 shows Mode 2 when receivers are no moving and moving at 0.9 (m/s), respectively.

First, obviously, the four diversity improved bit error rate significantly for both no moving and no moving cases.

Next, considering moving cases, as shown in Fig. 6 and 8, the blue triangle marked line BER is better than the black square marked line BER. This proved that ICI cancelling improved performance. In Fig. 8, blue triangle marked line is just slightly better than black
square marked line because OFDM symbol length of Mode 2 is shorter than Mode3, so Mode2 is more resistant to time varying channel.

Finally, the red line demonstrates that when freq.-IMP is applied, it also helps improve bit error rate. Fig. 9 will show more clearly improvement BER by applying freq.-IMP.

<table>
<thead>
<tr>
<th>Item</th>
<th>Content</th>
</tr>
</thead>
<tbody>
<tr>
<td>TX-RX Distance</td>
<td>51 m</td>
</tr>
<tr>
<td>RX Interval</td>
<td>0.3 m</td>
</tr>
<tr>
<td>Ocean Depth</td>
<td>3 to 5 m</td>
</tr>
<tr>
<td>Transducer Depth</td>
<td>1 m</td>
</tr>
<tr>
<td>RX Transducer Vel.</td>
<td>0 (stable), 0.6, 0.9 m/s</td>
</tr>
<tr>
<td>Weather</td>
<td>No wind</td>
</tr>
</tbody>
</table>

*Table 2: Experiment Parameters*

**Fig. 4: Experiment Site**

**Fig. 5: Mode3, QPSK, No Moving,**

**Fig. 6: Mode3, QPSK, Moving 0.6m/s**

**Fig. 7: Mode2, QPSK, No Moving**

**Fig. 8: Mode2, QPSK, Moving 0.9m/s**
In Fig. 9, received signal suffers impulsive noise, the red line means freq.-IMP is ON, and the blue line means freq.-IMP is OFF. At 13, 15, 17, 30 OFDM symbol point, impulsive noise is compensated successfully.

4. CONCLUSION

In this paper, an ultrasonic OFDM transceiver architecture with four diverse receivers, supporting Mode2 and Mode3, is proposed. Experiments were conducted to evaluate performance. To deal with challenges posed by the shallow sea transmission channel, we employed four techniques including diversity receiver, 2D-DFT channel estimation, ICI cancelling, time-IMP and freq.-IMP. As experimental result showed, those techniques have improved performance.

REFERENCES

PIC-DDFE-IDMA DETECTION FOR UPLINK SHALLOW WATER ACOUSTIC CHANNELS

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Abstract: In this paper, a new receiver structure is proposed for the uplink of interleaved division multiple access (IDMA) based transmission in shallow water acoustic channels. In a conventional IDMA rake detector, jointly removing inter symbol interference (ISI) and multiple access interference (MAI) is achieved by exchanging extrinsic log-likelihood ratio (LLR) chips between an elementary signal estimator (ESE) and a posteriori probability decoders (DEC) in a turbo-like manner. This principle can be extended to centralized decision feedback equalizer (CDFE) and DEC in CDFE-IDMA systems to offer improved performance. In contrast, the proposed IDMA receiver can be considered as a combination of a Parallel Interference C canceller (PIC) for eliminating MAI with a decentralized decision feedback equalizer (DDFE) to remove the ISI effects. The PIC utilizes the same ESE principles in removing MAI effects from received symbols, which is more efficient than cross-over filters in a CDFE-IDMA system. Moreover, the ISI subtraction in DFE is more effective compared to ESE. Moreover, by analyzing the computational complexity of the proposed PIC-DDFE-IDMA and comparing it with both rake IDMA and CDFE-IDMA on the basis of the required operations, it can be demonstrated that the PIC-DDFE-IDMA provides much lower complexity than rake IDMA detection, however, it is more complex than CDFE-IDMA. Utilizing experimental channels obtained by sea-trials conducted by Newcastle University in the North Sea, simulation results demonstrate the superiority of PIC-DDFE-IDMA over the other two detection methods.

Keywords: DFE, IDMA, DDFE-IDMA

1. INTRODUCTION

Interleave division multiple access (IDMA) is an alternative multiple access technique for wireless communication systems that can be used to replace code division multiple access (CDMA) based systems [1]. Rake IDMA [2] receiver is designed for multipath fading channels; however, it suffers from high complexity computations. Therefore, centralized decision feedback equalizer [3] IDMA (CDFE-IDMA) has been designed for two user underwater acoustic transmission which provides higher performance and lower complexity compared to rake IDMA [4]. Nevertheless, CDFE-IDMA has low efficiency in low Eb/N0 due to the delay in convergence of crossover filter taps. Thus, in this paper, a new IDMA receiver is proposed for underwater acoustic channels. The proposed receiver depends on the parallel interference canceller (PIC) and a decentralized decision feedback equalizer (DDFE) to remove multiple access interference (MAI) and inter symbol interference (ISI), respectively. Simulation results show robust system performance of DDFE-IDMA when it compared to CDFE-IDMA and rake IDMA detectors.
2. UPLINK IDMA SYSTEM MODEL

Fig.1 depicts the uplink IDMA transmitter of the \( k \)th user, where \( 1 \leq k \leq K \) and \( K \) is the number of users. Let \( \{d_k\} \) denote the binary message sequence and \( \{c_k\}_{i=0}^{N_dR_c-1} \) denote the user’s data after error control encoding with \( R_c \) representing the code rate and \( N_d \) denoting the number of transmitted data bits. A specific interleaver pattern for each user ( \( \Pi_k \) ) then permutes the coded output bits. These interleavers are generated pseudo-randomly and independently. After interleaving, each group of interleaved coded bits \( \{x_k\} \) are mapped to symbols \( \{s_k\}_{n=0}^{N_s-1} \), where \( N_s \) is the number of the transmitted symbols that are taken from a \( M \)-ary symbol alphabet: \( \chi \triangleq \{\alpha_1, \ldots, \alpha_M\} \) with \( E\{\chi\} = 0 \) and \( E\{|\alpha_q|^2\} = 1 \). Hence, the output symbols of each user \( \{s_k\} \) are preceded by a user-specific training sequence \( \{p_k\}_{q=0}^{N_t-1} \), where \( N_t \) is the number of the transmitted training symbols forming the transmitted frame.

We assume a channel model with \( L_k \) paths for the \( k \)th user, with complex-valued fading coefficients \( \{h_k(l)\}_{l=0}^{L_k-1} \). The received signal can then be represented as

\[
r(m) = \sum_{k,l} h_k(l)f_k(m - l) + v(m),
\]

where \( \{f_k\} \) is the transmitted signal frame consisting of a training preamble \( \{p_k\} \) and the interleaved sequence \( \{s_k\} \). Furthermore, \( m = 0, 1, \ldots, N_s + N_t - 1 \), and \( v(m) \) are samples of additive White Gaussian Noise (AWGN) with variance \( \sigma^2 = N_o/2 \). The received training and data symbols within \( r(m) \) in (1) can be written as

\[
r_t(q) = \sum_{k,l} h_k(l)p_k(q - l),
\]

and

\[
r_s(n) = \sum_{k,l} h_k(l)s_k(n - l),
\]

respectively.

3. ITERATIVE DDFE-IDMA DETECTION

The structure of the proposed DDFE-IDMA receiver is shown in Fig. 2. It generally includes of Parallel Interference Canceller (PIC), DFE and APP decoders. The main function of the PIC is to remove the MAI effects from the received signal belonging to a specific user.
The detection of the $s_k$ symbols from $r_s(n)$ in (3) can be rewritten as

$$r_s(n + l) = h_k(l)s_k(n) + \eta_{k,l}(n),$$  \hspace{1cm} (4)

where

$$\eta_{k,l}(n) = \sum_{k' \neq k,l} h_{k'}(l)s_{k'}(n) + v(n),$$  \hspace{1cm} (5)

represents the distortion, including residual MAI and ISI plus AWGN. The output of PIC for the $k$th user $s_k(n)$ is given by

$$e_{pic}(s_k(n)) = \frac{r_s(n) - E\{\eta_k(n)\}}{Var\{\eta_k(n)\}},$$  \hspace{1cm} (6)

where $E\{\eta_k(n)\}$ and $Var\{\eta_k(n)\}$ are the total mean and total variance of the interference, respectively. The decoders employ standard APP decoding on $\hat{d}_{dec}(c_k(i))$ to generate a posteriori log likelihood ratios (LLR)s $e_{dec}(c_k(i))$. The output of the DECs are the LLRs of $(s_k(n))$.

4. **COMPLEXITY COMPARISON**

The optimal values for filter taps could be easily implemented by utilizing adaptive algorithms. Normalized least mean square (NLMS) is a well-known and low complex algorithm that could be employed for that purpose. Table 1 summarized the total
The computational complexity of DDFE-IDMA, rake IDMA and CDFE-IDMA receivers for NLMS algorithm. One complex multiplication costs 4 real multiplications and 2 real additions. One complex addition/subtraction costs 2 real additions. For the given example shown in the table, it is obvious that CDFE-IDMA has a lower complexity than rake and DDFE-IDMA. This is because there is no complex interference cancellation in cross-over filters to remove MAI. Moreover, the absence of multiplying the channel conjugate with the received symbols, total mean and total covariance of the interference in DDFE-IDMA makes it having lower complexity than rake IDMA.

Table 1: Complexity comparison between rake IDMA, CDMA-IDMA, and DDFE-IDMA for K=8, L=16, it=8, N_f=N_b=N_c=16 utilizing NLMS algorithm.

<table>
<thead>
<tr>
<th>Operation No.</th>
<th>Rake IDMA</th>
<th>DDFE-IDMA</th>
<th>CDMA-IDMA</th>
</tr>
</thead>
<tbody>
<tr>
<td>Add./Sub.</td>
<td>ESE</td>
<td>NLMS-DFE</td>
<td>Example</td>
</tr>
<tr>
<td></td>
<td>22 × L × K × it</td>
<td>-</td>
<td>12620</td>
</tr>
<tr>
<td>Multiplication</td>
<td>46 × L × K × it</td>
<td>-</td>
<td>19512</td>
</tr>
<tr>
<td>Division</td>
<td>2 × L × K × it</td>
<td>-</td>
<td>782</td>
</tr>
<tr>
<td>tanh</td>
<td>2 × K × it</td>
<td>-</td>
<td>48</td>
</tr>
<tr>
<td>Total</td>
<td></td>
<td>32962</td>
<td></td>
</tr>
<tr>
<td>Add./Sub.</td>
<td>PIC</td>
<td>NLMS-DFE</td>
<td>Example</td>
</tr>
<tr>
<td></td>
<td>16 × L × K × it</td>
<td>(N_f+N_b) × 10 × it</td>
<td>8718</td>
</tr>
<tr>
<td>Multiplication</td>
<td>24 × L × K × it</td>
<td>((N_f+N_b) × 10+2) × it</td>
<td>12648</td>
</tr>
<tr>
<td>Division</td>
<td>2 × L × K × it</td>
<td>it</td>
<td>794</td>
</tr>
<tr>
<td>tanh</td>
<td>2 × K × it</td>
<td>-</td>
<td>48</td>
</tr>
<tr>
<td>Total</td>
<td></td>
<td>22208</td>
<td></td>
</tr>
</tbody>
</table>

5. SIMULATION RESULTS

The performance of both adaptive and optimal DDFE-IDMA detectors for two users are examined and compared in this section. The Eb/No is the bit to noise power ratio which is equal for both users. The simulation examples are using a 1/8 coding rate employing octal generators (275, 275, 253, 371, 331, 235, 313, 357)_8 that in turn determines the length of the interleaver schemes, which is 16000 chips in each frame. In addition, the channels are experimentally obtained by sea-trials conducted by Newcastle University in the North Sea for user1 and user2 as shown in Fig.3 and Fig.4, respectively. Furthermore, The length of 40 taps is chosen for feed-forward and feed-backward filters and 8 iterations are used for IDMA detection.

Fig. 6 illustrates the performance of DDFE-IDMA when NLMS and RLS algorithms are used. As it was expected, RLS is more close to providing optimal performance than NLMS; however, this is accomplished with greater system complexity. Fig.5 shows the
distribution of LLR for the feedback coded bits when RLS algorithm is used during IDMA iterations. Although, there is a great amount of overlap in first iteration, however, the quality of LLR coded bits gradually becomes better after three iterations. Eventually, after 8 iterations, the LLRs approach a bimodal Gaussian distribution implying that the ISI and MAI effects are removed by the DDFE and PIC, respectively.

Next, a comparison of the proposed receiver and the CDFE-IDMA, CDFE-CDMA and rake IDMA is depicted in Fig.7. From the figure, it can be seen that the detection performance achieved by DDFE-IDMA outperforms the other detectors. The absence of crossover filters in DDFE-IDMA provides higher efficiency in removing MAI effects.
1. CONCLUSION

In this paper, an iterative DDFE-IDMA detection has been proposed for two users in acoustic underwater channels. The utilization of DDFE for ISI abstraction and PIC for MAI elimination in DDFE-IDMA provides efficient detection against channel effects. Both of RLS and NLMS adaptive algorithms have been used for adapting the DDFE filters taps, and RLS algorithm was close to provide optimal performance. The improvement of the feedback LLRs of the coded bits within IMDA iterations exhibits the efficiency of PIC and DDFE in removing MAI and ISI respectively. Moreover, DDFE-IDMA showed to be more robust to channel impairments compared with CDFE-IDMA, CDFE-CDMA and rake IDMA systems, however, the complexity comparison depicted lower complexity of CDFE-IDMA receiver when it compared with DDFE-IDMA and rake IDMA detectors.

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Session 30

Underwater Unexploded Ordnance (UXO) Detection and Remediation

Organizers: Mike Richardson and Wolfgang Jans
FIRST RESULTS FOR BURIED OBJECT DETECTION FROM THE “SOUNDING AMMUNITION (SOAM)” 2013 EXPERIMENT

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Abstract: The German attempt to move away from fossil fuels and abandon nuclear power has put the focus on renewable energy, and especially on offshore wind farms in Germany. On the other hand, remains from World War II like dumped ammunition and other unexploded ordnance (UXO) in the North Sea and the Baltic Sea become in this context a more and more urgent problem. Currently marine magnetometers are often used to detect UXO at sea, although their range is very limited. Furthermore, some objects are made of alloys, making them undetectable by magnetometers. Hence, acoustic and electromagnetic methods can help with UXO detections. One part of the Sounding Ammunition Project (SOAM) is to compare different acoustic sonar systems. Therefore, several sea trials on a test site with submerged objects were planned. The test site contains several metal test objects. These objects are situated in different depths between flush buried and 1 m under the seafloor consisting of sand and muddy sand. The focus in the first sea trial 2013 was on the seafloor (side scan sonar, bathymetry) and on cross sections with two different sub-bottom profilers. A magnetic sensor was used to correlate acoustic and magnetic signals. This sea experiment will be described and first results will be shown. Additional, some prospects on measurements with different multibeam sub-bottom profilers and magnetic sensors will be made.

Keywords: Dumped Ammunition, Sediment Sonar, UXO, Modeling, SOAM,
1. INTRODUCTION

The planned exit from nuclear energy in Germany causes an increasing interest in offshore areas. The planned constructions of windmill farms, undersea cables and pipelines are making the problem of dumped ammunition and other unexploded ordnance (UXO) in the North and Baltic Sea very urgent.

The disposal of ammunition at sea was considered to be the safest and most efficient method until the seventies of the last century. Hence, dumped ammunition from the First and the Second World War causes today a big challenge for offshore companies not only in German waters but around the globe. Böttcher et al. [1] estimates that there are still up to 1,600,000 tons of conventional ammunition and up to 230,000 tons of chemical ammunition present within the German economic zone in the North and Baltic Sea.

The Sounding Ammunition (SOAM) project focuses on detecting unexploded ordnance in the sediment. The project is described in detail in [2] and aims at constructing an AUV mounted multisensor system with neuronal net based data analysis for UXO detection [3].

A test site set up by WTD 71 in the Baltic Sea has been used to collect data for the neuronal net based analysis and to compare the different acoustic and magnetic sensor systems.

This paper gives a short overview over the first experimental tests, it is organized as follows. Chapter 2 describes briefly the chosen test site and the buried objects. Chapter 3 gives a brief overview over tested and potential systems. Chapter 4 presents some preliminary results and in chapter 5 supporting modeling is presented for a selected system.

2. TEST SITE

2.1. Geology

Several glaciations episodes during the Pleistocene until 12,000 before present scooped out the Eridanos River to the Baltic Sea, where the main changes nowadays are due to sedimentation and caused by wind and waves [4, 5]. They erode material at the coast and in shallow waters and transport them into deeper areas. With increasing water depth and decreasing transport energy, the transported sediments are getting finer.

![Fig. 1: Schematic representation of the coastal area in the Lübeck Bay (modified after [5])](image-url)
The WTD 71 test site in the Lübeck Bay was set up 2003/2004 after several site characterization studies using side scan sonar, bottom samples and images from the sub-bottom profiler as well as videos from a remote operating vehicle (ROV).

The side scan sonar and the sub-bottom profiler are showing differences in the reverberation between the upper part in the east and the deeper part in the west.

Fig. 2: Sub-bottom image side scan from the investigation area, showing the surface and the subsurface with till, sand and mud [4]

2.2. Objects

In the years 2003 and 2004, seven objects were buried at different depths with a distance of twenty meters between adjacent objects. Those spheres (\(\phi = 60\,\text{cm}\)) and cylinders (\(\phi = 50\,\text{cm}, 180\,\text{cm}\)) are made of steel and filled with concrete with a density similar to the surrounding sediment.

An advantage of the spherical shape as buried object for testing detection methods is e.g. the independence of their target strength from angle.

The resent site verification almost ten years later shows similar results to former surveys. There are no indicators at the surface of the seafloor of buried objects.

Fig. 3: A core sample from the slope at the chosen site shows:

- 0 - 26 cm  Fine to medium sand with shells
- 24 - 40 cm  Medium sand with shells
- 40 - 139 cm  Decreasing sediment size with depth. Fine sand with remains of large plants
3. TESTED SYSTEMS

3.1. Magnetic

We measured e.g. during the sea experiment fall 2013 anomalies in the earth magnetic field, caused by objects with ferromagnetic properties. To do so, we used different types of magnetometers.

A G-882 Cesium-Vapor Marine Magnetometer from Geometrics has been applied as a single sensor 10 m behind our C3D-Sub-bottom Profiler and side scan sonar (Fig. 5).

Boskalis Hirdes EOD Services deployed similar sensors as a transverse gradiometer system. The sensors were towed about 3 m above the seafloor and linked to a differential GPS with ultra-short baseline communication (USBL).

A second type, a fluxgate magnetometer, is mounted on an AUV demonstrator system by Atlas Maridan. This AUV is built for pipeline inspections and uses eight FGM3D/100 UW sensors from Sensys.

Fig. 4: Bathymetry of the test site with unknown objects in the vicinity based on data measured by Boskalis Hirdes EOD Services

Fig. 5: C3D-SBP with G-882 Marine Magnetometer
3.2. **Innomar SES-2000**

The current survey of the sediment at the test site started with some profiles with a parametric sub-bottom profiler. We used the SES-2000 parametric sonar built by Innomar with a primary frequency of approximately 100 kHz and a resulting secondary frequency of approximately 6 kHz. The SES-2000 system was mounted in the moon pool of the vessel and equipped with a motion reference unit to remove heave, pitch and roll movement from the data.

During the measurements, we tried to pass the objects with different velocities to improve the lateral resolution. The -3 dB footprint at 18 m water depth is only 0.4 m, which makes mapping very difficult with a single beam system.

We measured some long profiles over the objects, to get an idea of the underground.

3.3. **Benthos C3D-SBP**

The Benthos C3D-SBP is a side scan sonar and bathymetry system (200 kHz) using the so called "Computed Angle of Arrival Transient Imaging" (CAATI) and a sub-bottom profiler (2 - 7 kHz). The towfish is equipped with some auxiliary sensors, like altimeter, a motion reference unit and the in 4.1 described G-882 magnetic sensor (Fig. 5).

3.4. **Sedison**

The Sedison is a low frequency multibeam echo sounder working at 10 - 20 kHz. It uses a Mills Cross antenna configuration with transducers perpendicular to the direction of movement. The receivers are inline. Both results in a good angular resolution.

Coherent overlapping of several pings is used to create a synthetic aperture and good lateral resolution.

3.5. **AUV mounted 3D Sub-Bottom Profiler**

The AUV mounted 3D sub-bottom profiler applies a low frequency sweep signal and a set of receiver pattern. The whole system and some results are described in [6] for pipeline inspection.
4. RESULTS

4.1. Magnetic

Figure 7 shows the girded results of the measurements on the test field, showing the anomaly of two spheres with a diameter of 60 cm in different depths. This data was collected by Boskalis Hirdes EOD Services using the Geometrics TVG-882 with a layback of around 65 m behind the vessel. A constant altitude less than 3 meters together with the high sensitivity of the magnetometers and a line spacing of 3 meters leads to very good results and an affirmation of the object positions.

![Fig. 7: Magnetic field anomaly measured by Boskalis Hirdes EOD Services over two spherical objects in different depths](image)

The measured total field of a single sensor is shown in figure 8. The side scan sonar image is plotted together with the signal of the total magnetic field for a single track line.

![Fig. 8: Side scan sonar and magnetic field anomaly over two spherical objects in different depths](image)
4.2. Sub-Bottom Profiler

The raw data image (Fig. 9) from the sub-bottom profiler SES-2000 shows a signal for a sphere with 60 cm diameter.

To account for the problems with the single beam system, the measurements are expanded to a multi transducer system, based on the SES-2000 Quadro by Innomar.

![Figure 9: Example of SES-2000 raw data with hit on the 60 cm sphere](image)

5. MODELING

Measured results are compared with simulations. An example is given for the sub-bottom profiler SES-200. For simulating the signal-to-noise ratio of the sphere the numerical simulation tool MSM (Mine hunting Simulation Model) is used and an example is shown in figure 10. Figure 10b exhibits the echo level of a sphere and the noise components for three different depths of burial versus slant range (equivalent to travel time). It shows that the detection with a sufficiently high SNR is possible using a parametric sonar and gets more difficult for flush buried objects.

![Figure 10: a) SNR at 6 kHz for a sphere of steel (60 cm) filled with light concrete (Target strength: -16 dB) b) Echo level and Noise for this sphere at 10 cm, 20 cm and 30 cm burial depth](image)

6. SUMMARY AND OUTLOOK
Within the SOAM project first sea experiments were carried out. The characterization of the chosen test site is given in this paper and first results for selected magnetic and acoustic sensors were presented.

The comparison of different acoustic and magnetic sensors will be expanded in the near future. The goal is to select a set of sensors for an AUV, which fit best for the problem of detecting dumped ammunition in the sediment.

7. ACKNOWLEDGEMENTS

The Project SOAM is coordinated by Clausthaler Umwelttechnik-Institut GmbH. Further project partners are the Research Department for Underwater Acoustic and Marine Geophysics (FWG) of the Bw Technical Center for Ships and Naval Weapons, Maritime Technology and Research (WTD 71), Atlas Elektronik and Challenger Technologies. Special thanks to the project partner Boskalis Hirdes EOD Services for the collected data. The SOAM project is founded by the Federal Ministry for Economic Affairs and Energy within the TIMM program managed by the Projektträger Jülich (Grant FKZ: 03SX341).

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TARGET DETECTION WITH SECTOR SCANNING IMAGING SONARS

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Abstract: The probability of detecting targets of interest on the seafloor with imaging sonars depends on the spatial resolution and the contrast of the image. This paper presents a "proof-of-concept" demonstration that commercial sector scanning imaging sonars installed at a fixed location can be used effectively to detect targets of interest in a scene under surveillance. In general, such sonars output beamformed time series of acoustic backscatter magnitude for each ping, with little or no corrections applied to the data. Our approach uses acoustic backscatter data acquired over the entire scanned sector, and over a time span much longer than the temporal coherence time of the scene, to estimate and then remove, the deterministic spatial gain introduced by the combined transmit and receive beam patterns. In the process, measured backscatter values are normalized for the size of the area ensonified by the pulse at each range increment within the beam, with compensation applied for transmission losses incurred during the round-trip path between the sonar and the seafloor. Targets that are difficult to discern in a raw image become clearly visible after removal of the beam pattern signature and normalization of the backscatter values. Such corrections should lead to improved target detection by sonar operators and by automatic target detection and recognition algorithms.

Keywords: target detection, change detection, sector scanning imaging sonar, beam pattern estimation
INTRODUCTION

Acoustic imaging sonar technologies, including very sophisticated military-grade mine-hunting sonars, have been used for decades to detect munitions on the seafloor. These technologies are well suited for detailed seafloor surveys in potentially contaminated underwater areas, but they are often hampered by various fluctuating environmental factors such as suspended sediments, fish, migrating bedforms, background noise, and transmissions from other nearby sonars. In addition, targets of interest may be obscured by changes in echo levels imparted by the beam pattern of the sonar.

A variety of low-cost imaging sonars have also been used to find munitions underwater. In general, these sonars output beamformed time series of acoustic backscatter magnitude for each ping, with little or no corrections applied to the data. Although it is desirable to calibrate a given sonar system in a controlled environment before deploying it in the field, it is often impractical to account for the exact mounting geometry and for the environmental conditions in which the sonar will operate in the field. In fact, the same transducer array will exhibit notable differences in the shape of its beam pattern (e.g. relative position of sidelobes) for different mounts and in different environments. Consequently, it is necessary to develop a method to estimate beam pattern signatures in situ, using the data being collected.

Using data collected during an entire survey performed with a seafloor swath mapping multibeam echosounder mounted on a moving platform, de Moustier and Kraft (2013) developed a technique to determine the composite transmit and receive beam pattern signature of the sonar from acoustic backscatter. The process converts raw acoustic backscatter time series output by the sonar to an acoustic backscatter time series with intensity levels that are independent of the characteristics of the measuring sonar and independent of range. The same concept was applied here to a sector scanning sonar mounted on a fixed platform. A sector scanning sonar transmits and receives on a single beam, which is rotated mechanically to cover a polar field of view. Each angle step in the scan represents an independent realization of the beam pattern against a different seafloor background. Over time, the overall scene is likely to change as sediments are mobilized by wave and current activity. An average of many realizations over time will yield an estimate the beam pattern, which represents the deterministic component of the recorded time series.

In the following we demonstrate a “proof-of-concept” for the beam pattern estimation technique using acoustic backscatter data recorded with a sector scanning sonar. The sonar was mounted on a quadpod instrument tower (Fig. 1) deployed in 20 m water depth offshore of Panama City Beach, Florida, during the Target and Reverberation Experiment 2013 (TREX13) sponsored by the U.S. Office of Naval Research. As part of the deployment, several machined targets shaped like munitions of various calibers (Fig. 2) were placed on the sandy seafloor in the field of view of the sonar. The goal was to quantify target mobility and seafloor roughness while concurrently acquiring time series of waves, currents, and suspended sediment concentrations with other instruments mounted on the quadpod.
Fig. 1: Imagenex 881 Tilt Adjust imaging sonar mounted on one of the legs of a quadpod tower. For scale, the red pressure case of the sonar (white arrow) is 320 mm long and 89 mm in diameter.

Fig. 2: Diver photo of targets placed on the seafloor. The target characteristics are listed in Table 1. For scale, the target diameters range from 155 mm to 20 mm (A - D).

METHOD

Targets were fabricated using solid materials (e.g., delrin plastic and stainless steel) to have overall dimensions, shape, bulk density, and rolling moments similar to munitions of interest. The targets deployed have a stark difference in their acoustic and electromagnetic response when compared to their real counterparts; however, we expected their mobility characteristics to closely resemble those of their real counterparts. General target properties are listed in Table 1.
The sector scanning sonar operated at an acoustic frequency of 2.25 MHz with a nominal beam width of $1.7^\circ \times 30^\circ$ (azimuth $\times$ elevation). The sonar was set to transmit a 20 µs continuous wave (CW) pulse, and to scan a 102º azimuthal sector in 0.3º increments out to a maximum range of 6 m from the sonar head. Echoes were sampled at 250 kHz for a nominal range sampling interval of about 3 mm, which is roughly 5.7 times the nominal range resolution of the 20 µs CW pulse. A complete 102º scan comprising 340 pings was recorded every 12 minutes, with every twentieth scan skipped (i.e., 19 scans every 4 hours) repeating for 33 consecutive days. During the deployment, waves and currents associated with local storms fully reworked the seafloor scene being imaged.

The altitude of the transducer array phase center was 1.27 m. Data from a conductivity, temperature and depth (CTD) sensor installed on the quadpod tower provided continuous monitoring of the in-situ acoustic absorption coefficient needed to compensate for transmission losses. At 2.25 MHz, and in 20 m of water depth with a relative salinity of 35, absorption decreases from about 1.62 dB m$^{-1}$ at 10 ºC to about 1.21 dB m$^{-1}$ at 20 ºC. Therefore, as temperature changes, absorption alone can account for variations of 1-2 dB in the received backscatter levels.

<table>
<thead>
<tr>
<th>Label</th>
<th>Diameter (mm)</th>
<th>Length (cm)</th>
<th>Volume (cm$^3$)</th>
<th>Mass (kg)</th>
</tr>
</thead>
<tbody>
<tr>
<td>A</td>
<td>155</td>
<td>58.4</td>
<td>7683.9</td>
<td>34.2</td>
</tr>
<tr>
<td>B*</td>
<td>81</td>
<td>52.5</td>
<td>1209.4</td>
<td>3.8</td>
</tr>
<tr>
<td>C</td>
<td>25</td>
<td>21.9</td>
<td>165.5</td>
<td>0.39</td>
</tr>
<tr>
<td>D</td>
<td>20</td>
<td>16.8</td>
<td>77.0</td>
<td>0.20</td>
</tr>
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Table 1: List of replica targets fabricated from solid materials to have size, shape, bulk density, and rolling moment similar to munitions. *Target B contained fins.

RESULTS AND DISCUSSION

An example of a complete scan (340 0.3º steps) of the scene with targets is shown in Fig. 3a. Each radial line in the image corresponds to one ping (one 0.3º step). The acoustic backscatter values have been corrected for transmission loss and they are displayed on a logarithmic scale in decibels (dB) relative to an arbitrary reference. The 22-dB spread in the backscatter values is significant, not the values themselves.

A sonar operator may discern some targets in the “raw” image between 2 m and 3 m from the origin (Fig. 3a), and probably cannot identify the other elements of the picture as seafloor features or targets. Likewise, an automated target detection algorithm would struggle to extract targets from such an image because of (1) the changes in backscatter levels with range due to the beam pattern of the sonar, and (2) the corresponding large (>20 dB) dynamic range of the signals.

By contrast, a sonar operator should have no trouble identifying at least 7 targets in the corrected image shown in Fig. 3b with the estimated beam pattern removed. An automated detection algorithm will also be much easier to train on such an image with an intensity dynamic range of about 5 dB, and in which acoustic shadows (low backscatter) are well defined behind each object that cast them. In addition, the intensity level at the onset of the shadows is nearly independent of the range of the object from the origin. Likewise, the intensity level of the seafloor acoustic backscatter displayed in the image should be range independent for a given bottom type. As in Fig. 3a, only the 5-dB spread in the backscatter
values in Fig. 3b is significant, not the values themselves. If the 2.25-MHz target strength of at least one of the targets in the scene were known, it could be used as a reference for the entire image and the dB scale would no longer be arbitrary.

Fig. 3: Beam pattern signature removal in seafloor images obtained with a sector scanning sonar operating at an acoustic frequency of 2.25 MHz. Acoustic backscatter image (relative dB scale) for (a) a single uncorrected scan, and (b) with beam pattern removal and temporal averaging. Labels placed to the right of the seven targets identified in the image refer to target characteristics in Table 1.

With a nominal azimuthal beamwidth of 1.7°, there is over 82% overlap between adjacent beams in the scan. However, the 0.3° offset between adjacent beams was sufficient to provide an independent realization of the beam pattern with a different seafloor contribution. Before averaging many realizations over scan angles and scan times, the acoustic backscatter values were normalized at each range increment by the size of the area ensonified by the 20 µs pulse within the azimuthal footprint of the 1.7° beamwidth. For this area computation, the seafloor was approximated as a horizontal plane over the 6 m range.
extent considered here. The geometry of this area computation is similar to that described in de Moustier and Alexandrou (1991).

The data set used for this demonstration suffers from receiver saturation at slant ranges between 1.5 m and 3 m, which can be seen as a band of high backscatter in Fig. 3a and a band of overly smooth seafloor surface in Fig. 3b. The beam pattern cannot be fully estimated for the elevation angles corresponding to this band of saturated returns, yet the corrected image is still improved with respect to target detection.

The particular sonar used in this experiment applied a time-varying gain to the received echoes to compensate for transmission losses. The gain function was implemented as a sequence of 1 dB steps. However, there was a spike at the transition from each step to the next. Unfortunately, the spikes were not regular enough to be removed with the beam pattern, so they remained in the images and appear as concentric rings (Fig. 3).

CONCLUSIONS

We have shown that it is possible to obtain an in situ estimate of the combined transmit and receive beam pattern of a sector scanning imaging sonar deployed at a fixed location for 33 days. The beam pattern was estimated by spatial and temporal averaging of scan lines. Before averaging, the acoustic backscatter values were normalized for the size of the area ensonified on the seafloor by the transmitted pulse at each range increment. The estimate of the combined transmit and receive beam pattern must be obtained over time scales much longer than the temporal coherence time of the scene. The underlying assumption of the approach is that the scene will remain globally invariant over days to weeks during calm weather conditions, or over months before and after a major storm. Therefore, relative to the global norm, discrete targets appearing in the scene will be readily detectable and straightforward to localize at the spatial resolution of the sonar. Nonetheless, even the best signal processing tools cannot correct for data acquisition issues such as signal saturation or insufficient receiver gain. Likewise, the signal processing will be compromised by missing environmental parameters such as time series of temperature, pressure, and salinity needed to compute absorption, a critical parameter in change detection at acoustic frequencies above 100 kHz.

ACKNOWLEDGEMENTS

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REFERENCES

LOW FREQUENCY FEATURE EXTRACTION FOR TARGET DISCRIMINATION ON A BIOSONAR DATASET

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Abstract: Low frequency broadband sonar is an emerging technology in the field of underwater unexploded ordnance (UXO) detection and identification. These lower frequency acoustic signals can penetrate an object and allow one to extract target features based on both elastic effects and internal structure that will allow for more accurate target classification. One such sonar, Hydrason’s (UK) Biosonar is a broadband side-scan sonar that operates between 30 kHz and 130 kHz and uses dolphin inspired pulses. This paper examines data from trials conducted by the UK with Biosonar in the fall of 2012. Particular focus is given to the classification of a series of cylindrical targets with identical structures yet differing internal fillers. For these targets, numerous features are considered in order to differentiate between the targets i.e. their fillers. These features include time-domain features, spectral-based features, and aural-based features. Simple classification schemes are applied and the suitability of the features is discussed in the context of an ATR (automated target recognition) scheme.

Keywords: Automatic Target Recognition, Low Frequency Sonar, Aural classification, UXO
1. INTRODUCTION

In the field of underwater unexploded ordnance (UXO) detection and identification, the current state of automatic target recognition (ATR) relies on high-frequency imaging sonars. The use of ATR with these sonars, while powerful, is susceptible to high false alarm rates in areas of high clutter. This is particularly true when clutter objects possess similar dimensions to the targets of interest. Low-frequency (LF) broadband sonar is a rapidly emerging technology in this field because of its potential to generate acoustic returns from both the surface of a target and its internal structure [1] and thus mitigate the problem of false alarms.

There has been some previous research into ATR with LF broadband sensors. The Naval Research Laboratory recently completed a study [1] where they generated unique “target fingerprints” – 360° target-aspect vs frequency plots – for a variety of UXO targets in the 1 kHz to 140 kHz frequency band. They further tested several classification schemes on the energy spectrum of these targets with good results. Significant work on understanding and accurately modelling the complexities of proud target scattering at lower frequencies is being conducted at the University of Washington’s Applied Research Laboratory [2]. Using the San Diego sonar system – a broadband sidescan sonar operating at 70 kHz to 120 kHz – Pailhas et al. [3] recently demonstrated the ability to consistently locate the resonance notches in a target’s echo spectrum. The locations of these notches can be thought of as a partial image of a target’s “fingerprint”.

2. DATA PROCESSING

During October 2012, Hydrason and Atlas Elektronik UK under contract to DSTL, UK conducted trials in Weymouth Bay using Hydrason’s Biosonar system. This sonar operates between 30 kHz and 130 kHz and has a range of greater than 50 m. During the trial, the Biosonar was mounted on a REMUS 100. This permitted the use of the vehicle’s native Marine Sonic sidescan sonar to generate co-located high-frequency images for each target survey. The vehicle was equipped with both a port and starboard Biosonar. Thirty missions were completed using the AUV in this configuration and this work examines two of the missions designated Mission 6 and Mission 7.

In both of the missions considered here the AUV acquired and then circled a series of nine gas cylinders oriented at varying target aspects along a north-south path. In Mission 6 the targets are ensonified by the port-side sonar and in Mission 7 the targets are ensonified by the starboard-side sonar. It is worth noting that there is a difference in centre frequencies between the two sides. The portside has a centre frequency of 60 kHz while the starboard-side has a centre frequency of 90 kHz. The vehicle had an average altitude of 3 m (in approximately 15 m water depth) and circled the cylinders with a radius of approximately 15 m throughout both missions.

Each cylinder contained one of three fills, water, kiln-dried sand, or gravel. As each target is structurally identical, this paper seeks to discriminate between the fill types. A fourth target arises from the concrete sinker used to anchor the water-filled cylinders. Fig. 1 shows an example of the targets in question along with the typical Biosonar “image” for a gas cylinder.
The authors found it instructive to represent the Biosonar returns in a manner which would be visually intuitive. Using the navigation data generated by the Marine Sonic sonar, the Biosonar echoes were aligned and polar plots of the match-filtered data were generated. Fig. 2 shows two of these plots for the gravel cylinders. As one might expect, there are differences arising from the differing orientations of cylinders of the same fill type. However, it is worth noting just how much these images vary for identical fill types. There is also a high level of noise in some parts of the data. The challenge in the classification problem is to find some set of features, temporal and/or spectral, which are consistent for a target type yet are discriminatory between the different target types.

3. SINGLE PING ANALYSIS

Once the target echoes were located, various sections of the matched-filtered sonar data were manually cropped from the wideband images of the cylinder. Within these cropped regions, the data was further reduced by considering only the section of pings within the
region whose maximum value exceeded a specified threshold. In addition, the maxima of these pings had to exceed the median value of the cropped section by a factor of 4.

These echo time series were approximately aligned using cross-correlation between adjacent pings starting with the ping with the largest amplitude return. For the ping with the biggest return, there could be earlier returns which exceeded the specified threshold. The first such occurrence was taken to correspond to the time of t=0 ms in the new echo time history and the other pings were aligned according to the cross-correlation time shift. The new set of time shifted pings had a length of approximately 3 ms (2600 points). In some cases, the pings were zero-padded at either the beginning or the end of the image if the start or end point of the extracted ping lay outside the original wideband image.

The number of pings considered was then further reduced by specifying a threshold on the maximum ping value. An example of the final aligned pings for Missions 6 and 7 are shown in Fig. 3. As can be seen, the pings have been well-aligned. The results from the 2 missions are similar although Mission 7 appears to have more background reverberation. This is likely due to the seabed being composed of broken shells in mud and the higher frequencies of the starboard sonar. Although the ping returns are centred about zero time, there is, in general, significant energy before and after as well. For example, at the top of Fig. 3a there is energy arriving at about the 1.5 msec time.

For each mission, or combined mission, the mean time series is computed. This mean is then subtracted from each time series and a “P” matrix is constructed with dimensions of npings x 2600. The eigenvectors of PTP then form the principal components of the echo series. The number of features can be reduced from 2600 to NP, the number of principal component coefficient values. The optimal combination of this reduced set of features that will discriminate between the 3 target classes can be determined using the method of Fisher discrimination [4]. In the case of 3 classes, there are 2 linear combinations of the NP principal components, which optimally (assuming Gaussian distributions of features) separate the classes.

In the following analysis, Missions 6 and 7 are considered separately using 2 thresholds on the amplitude of the magnitude of the match-filtered data – first a threshold of 8 and then a threshold of 12. In all cases, 80 principal components are used. In Fig. 4, the resulting distributions for the 3 classes are shown, with blue representing Class 1 (sand), red Class 2 (gravel), and green Class 3 (water). As can be seen, there is good class separation for the
water-filled cylinders. The sand and gravel are less well-separated, but one can use only some regions of the feature space for making a decision. In other words, if the feature values for a ping lie within a region with significant class overlap, do not use that ping for classification purposes. The data from Mission 7 appears to have better classification potential than the Mission 6 data. The threshold 12 data from Mission 7 has the best class separation.

![Feature distributions for three classes: blue–Sand, red–Gravel, and green–Water. The top row shows Mission 6 with thresholds of 8 and 12. The bottom row shows the respective distributions for Mission 7. Features 1 and 2 are the linear combinations resulting from the Fisher discrimination method.](image)

Thus far, the Principal Components and the Fisher Discriminant features have been computed using all the data from a mission. One can also consider taking the data from the first 2 sets of water, sand, and gravel targets to construct the features and then use the third set to investigate the class separability. This is considered for the Mission 7 data and the resulting feature distributions are shown in Fig. 5. As can be seen, the distributions of Fig. 5b which are based upon the constructions from Fig. 5a are less separable than the in Fig. 5a, but class separability does exist.

The classification results obtained using the ping features for Mission 6 are now shown in Fig 6. First the Fisher feature distributions with a threshold value of 12 (Fig. 4) are computed and then the thresholds on the values of Feature 1, $f_1$, and Feature 2, $f_2$ are applied using the rules:

- $f_1 < -0.1$ Water,
- $f_1 > 0$ & $f_2 > 0.05$ Sand, and
- $f_1 > 0$ & $f_2 < -0.05$ Gravel.
The resulting ping classifications along with the true labels are shown in Fig. 6. The label zero corresponds to the ping being a feature region not satisfying the rules outlined above. As can be seen, the resulting ping classifications are not always correct; however, for each instance of a target, the majority of the pings appear to correspond to the correct classification. Finally, the Fisher discriminant coefficients (the Principal Component coefficients) can be used with the Principal Components to give an interpretation of the Fisher features as projections on the 2 time series shown below in Fig. 7.

4. AURAL FEATURE ANALYSIS

Perception-based (aural) signal features have been shown to improve the classification performance of active sonar in highly cluttered areas [5]. The perception-based feature set is a representation of how the human ear perceives the timbre of a sound. The features have been employed in the Anti-Submarine Warfare community for target tracking, and in the oceanography community to discriminate between cetacean vocalizations and have proven quite robust [5, 6].
The methodologies for calculating the aural feature set are well laid out in the paper by Young and Hines [5] and only the major differences will be noted herein. As the aural features are a representation of how the human ear perceives timbre, the method deals with the range of audible frequencies, 20 Hz – 20 kHz. To address this, the straightforward approach of scaling the frequencies of the features up to the Biosonar operating range was chosen, thus preserving the entire bandwidth of the dataset. Previous work with the feature set employs Butterworth filters to improve the signal-to-noise ratio [5], however the data was match-filtered instead.

A total of 32 features were calculated for each of the four target types and after some analysis five were chosen. The first two features arise from the statistical breakdown of the subband decay, a time-frequency quantity. They are the frequency corresponding to the minimum of the subband decay slope, minSBDS-F, and the frequency corresponding to the maximum of the subband decay time, maxSBDT-F. The remaining three features are purely spectral and are the loudness centroid, LC, the peak loudness value, PLV, and the frequency corresponding to the peak loudness value, PLF. More comprehensive definitions of the features can be found in the paper by Young and Hines [5].

Once again, the method of Fisher discrimination was employed to determine the optimal combination of the chosen features. The resulting class separations are shown in Fig. 8 as the projections onto the three feature planes rather than as a 3D image. The gravel-filled cylinder and the sand-filled cylinder don’t demonstrate any apparent class separation. However, there is good class separation between the water-filled cylinder, the sinker.

5. DISCUSSION OF RESULTS AND FUTURE WORK

In this paper, only a small portion of the trial data set was considered and in future the authors will expand their approach to consider data from other portions of the trial. In the single ping analysis, it has been shown that the water, gravel, and sand target classes appear to be separable in feature space. As one might expect, the water class appears most separable from the other 2 types of targets, but is does seem possible to also discriminate between the gravel and sand classes using these features. The Aural feature analysis demonstrated that the water and the sinker classes separate the best, while the sand and gravel classes are indistinguishable from one another. In future work, a more rigorous training/testing of the classification algorithms will be considered. The circular plots also suggest that the classification problem might be treated as a pattern recognition problem.
and the authors would like to consider methods such as SIFT/SURF features or template-matching.

Figure 8: Class distributions using the method of Fisher discrimination for four classes: blue–Sand, red–Gravel, green–Water, and cyan–Sinker. Beginning from top left and proceeding clockwise the figures demonstrate the class separations in the 1-2 plane, the 1-3 plane, and the 2-3 plane.

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STATE OF THE ART IN COMMERCIAL OFFSHORE UNEXPLODED ORDNANCE DETECTION

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Abstract: Millions of tons of explosive remnants of war (ERW), including both conventional and chemical stockpile munitions, have been dumped in maritime environments worldwide. Also, military exercises, testing and warfare at sea have left behind large quantities of unexploded ordnance (UXO) at sea. In the North Sea and the Baltic alone, approximately 700,000 mines were laid in the First and Second World Wars, most of which were never recovered. With the increasing utilization of the maritime environment for energy production (offshore oil and gas, offshore wind energy, offshore tidal power), international trade (harbor construction and extension), and the production of maritime food, the clean-up of UXO and dumped conventional and chemical munitions becomes more and more important. Compared to the year 2010, the market for offshore UXO detection and removal has multiplied.

Standards for offshore UXO detection and removal as they have been adopted in land applications remain yet to be developed. Because of a lack of experience on the side of service providers – many of which are new to this market – and due to a lack of awareness for the dimension and risks of the offshore UXO problem on the side of the clients, many surveys carried out fail to detect the UXO sought. Careful consideration of the characteristics of state-of-the-art detection systems and objects of interest and environmental parameters allows to design UXO surveys such that the objectives are met. Systems applied include scalar and vectorial magnetometers, metal detection systems, sidescan sonar systems and high resolution multibeam echosounder systems.

Keywords: Unexploded Ordnance, UXO, Survey, Detection, Magnetometers, Sidescan Sonar, Multibeam Echosounder
1. THE OFFSHORE UNEXPLODED ORDNANCE PROBLEM

Millions of tons of explosive remnants of war (ERW) are resting in the seas worldwide. ERW are made up of dumped munitions (both conventional and chemical) and unexploded ordnance (UXO) from warfare, military training and exercises. The North Sea, the Baltic, the Irish Sea, the Skagerak, the English Channel with the coasts of Normandy and others [1, 2].

Offshore ERW and UXO are a world-wide, international problem. Munitions are found near beaches, in tidal areas and far offshore. In the North Sea and the Eastern Atlantic alone more than 150 munitions dump sites have been identified [2]. That number does not include former minefields and unexploded ordnance originating from naval warfare, military training and exercise.

Despite considerable marine areas being affected and large quantities of ERWs and UXO disposed of or lost in marine environments, offshore UXO detection and removal operations were required on a very small scale only until about 2010.

Since then, because of an increasing utilization of the marine environment in particular in coastal areas and along the continental shelfs for economic purposes, offshore UXO detection and removal has been experiencing a boom. This has been caused in particular by the installation of numerous offshore wind farms in shallow waters along the coasts of Europe.

The hazards and risks posed by ERWs and UXO for marine construction projects must be assessed based on historical information (archive research and literature review) on naval mine fields, dump sites, marine warfare and military practice and exercise grounds to assess probability to encounter ERW and UXO and the types that may be encountered. Depending on the probability of encounter and the hazards that can be outlined based on types, net explosive content, prevalence of sensitive or dangerous fuzes, water depths and planned construction activities, the necessity for UXO survey and investigation / removal of potential ERW / UXO objects must be determined in a risk assessment. However, it must be stated that while historical research can yield valuable information on potential ERW and UXO in a project area, historical research can never yield proof that allows to exclude ERW and UXO risks because many dumping and warfare activities were not documented well if at all.

Based on the risk assessment the survey objective must be defined in terms of a certain minimum size (ERW / UXO) object. Depending on the results of the risk assessment this may either be a certain type of ERW / UXO made from steel, a certain type of ERW / UXO made from nonferrous material, or simply an object of a certain size without material specification. Together with local seabed conditions (solid / rocky without mobile sediments or mobile sediments), these parameters are the starting point for UXO survey design.

Survey objectives in recent offshore and nearshore UXO surveys prior to dredging operations, cable laying or offshore wind farm installation range from 105 mm antiaircraft artillery (AAA) shells containing as little as 0.25 – 1.3 kg of explosives and “50 kg steel objects” to German LMB aluminium ground mines containing almost 700 kg of high explosives. Common survey objectives are “50 kg ferro” and “100 lbs bombs”.

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2. OFFSHORE UXO DETECTION

Detection tools for onshore (land) UXO detection have been developed over many years and have been standardized to a high degree. Typically, magnetometers are applied to detect ferrous (iron, steel) objects which could be UXO. Alternatively, metal detection systems are used to detect ferrous and non-ferrous metal objects. Both methods require the excavation of all targets detected above a certain threshold or size depending on the UXO clearance objective. Depths of detection for magnetometers have been proven to range from a few centimetres or decimetres for small calibre (infantry, small calibre artillery) to few meters for large unexploded bombs (e.g. ca. max. 2 m for 8.8 cm artillery shells or max. 4 – 5 m for 500 / 1,000 lbs unexploded bombs) in areas free from clutter objects and background noise. For conventional metal detection systems, the depths of detection are typically even less.

However, in offshore UXO detection, many service providers competing for market shares claim much larger depths of detection. Often, magnetometer surveys for unexploded ordnance in offshore projects are realized with magnetometer altitudes of more than 5 m above seabed and line spacing between adjacent survey lines of 10 m to 50 m and more. It seems justified to question why unexploded ordnance of similar type and size would be more readily detectable in offshore environments than in land applications, supposedly allowing much higher distances from the ground and much larger line spacings.

SURVEY INSTRUMENTS AND METHODS

UXO surveys in water bodies can be conducted with the following geophysical instruments:

- Magnetometers (scalar and vectorial magnetometers)
- Pulse induction metal detectors
- Sidescan Sonar Imaging Systems
- Multibeam Echosounder Imaging

Subbottom imaging systems have been tested for UXO detection but were not yet proven to provide the detection characteristics and reliability required for UXO detection.

Magnetometers can be used to detect anomalies caused by ferrous (mainly iron / steel) objects in the Earth magnetic field, while metal detectors can be used to detect any kind of (electrically conductive ferrous and non-ferrous) metal. As geophysical systems which are based on static (DC) magnetic field and electromagnetic field measurements, magnetometers and metal detection systems are rarely affected to a significant degree by the sediments or background geology. Both systems can be used to detect objects buried in the seabed.

Sidescan sonar and multibeam echosounder imaging systems are acoustic imaging systems that are used to acquire reflection patterns (acoustic backscatter), thus allowing to distinguish objects and structures on the seabed depending primarily on the spatial
resolution of the survey. Side scan sonar and multibeam echosounder systems cannot be used to detect objects buried in the seabed.

SURVEY DESIGN

When designing an offshore UXO survey, it is important to consider the physical properties of the survey objectives and the environment in which the survey has to be carried out. Based on this consideration, it can be determined if and which physical contrast between the survey objective and the “survey background” is most promising to achieve the survey objective, i.e. the reliable detection of all objects defined as survey objectives. The following examples illustrate the task:

Object 1: The German 105 mm AAA shell “10,5 cm Panzergranate Flak” measures ca. 390 mm in length, has a gross weight of ca. 15.5 kg, contains about 0.25 kg of high explosives and is made from steel.

Object 2: The US 100 lbs aircraft bomb “DEMO 100lb M30” has a diameter of ca. 210 mm, a length of ca. 760 mm and a gross weight of approximately 48 kg. It contains about 24 kg of high explosives in a steel casing.

Object 3: The German “LMB” (Allied World War II designation “GC”) ground mine has a diameter of 660 mm, a length of 2641 mm and a gross weight of circa 990 kg. In its aluminium housing, it contains approximately 700 kg of high explosives.

Object 1 is small, may be buried in sandy sediments or lie on a rocky or gravel seabed between gravel and pebbles of similar size. Although it is made from steel, the magnetic field anomaly caused by this type / size of shell can be assumed to be relatively small.

Object 2 is of medium size, may still be buried in sediments but is likely to stand out in terms of size and geometry among other objects on a rocky or gravel seabed. The magnetic field anomaly caused can be expected to be considerably larger than that of the 105 mm AAA shell.

Object 3 is large and unlikely to be completely buried in sandy sediments except where underwater dunes occur. Because of its size and geometry, it will clearly stand out from other objects on the seabed even on rocky bottoms. However, it cannot be expected to exhibit a significant magnetic field anomaly because it is made from nonferrous material (aluminium).

All three objects are made from metal which is conducting electricity, allowing to detect the objects with pulse induction metal detection systems.

Depending on the geological background in which the search for UXO is to be carried out, different methods or method combinations may be used to achieve the survey objective.

For example, in a magnetically neutral background geology with a mobile sediment layer of up to one meter, both magnetometers and metal detectors could be used to detect objects 1 (105 mm AAA) and 2 (100 lbs bomb). However, only the metal detector system would be able to detect object 3 if it was completely buried in the sediments. Side scan sonar and multibeam echosounder imaging would not be able to detect buried objects but should still be considered as auxiliary sensors to detect relevant objects lying on the seabed and to distinguish abundant scrap metal objects from potential UXO targets. While basic assessments of object size are possible with both magnetometer and metal detection systems, both methods do not allow to exclude objects as non-UXO. This is only possible
if during data interpretation other sensor information such as acoustic sensors are used to acquire additional information on the same objects.

In another case where the background has a strong magnetic influence (e.g., granite rocks / boulders or channels with fine silt material), magnetometers may produce data in which relevant (potential UXO) objects cannot be distinguished from background geology signatures. In these cases, only metal detection systems will allow to detect objects buried in sediments.

On solid (rock, gravel) seabeds, sidescan sonar and multibeam echosounder systems could be the most suitable sensor systems for the detection of large aluminium UXO such as object 3 (aluminium ground mine).

Another factor in the selection of survey systems is productivity. While multibeam and sidescan sonar imaging provide data fastest and at the lowest costs, magnetometers – even when applied in multichannel configurations – require much more time if a dense line spacing as required for reliable detection is used. Metal detector surveys require even more time because large-diameter loops need to be guided close to the seabed and 100% coverage must be obtained to ensure reliable detection.

**MAGNETOMETER SURVEY**

All ferrous objects are subject to two types of magnetization, relatively consistent induced and highly variable permanent magnetization. Both are vectors that add up to determine the resulting magnetic field of an object. Because the vector of permanent magnetization is fixed in the object and induced magnetization varies with orientation for non-sphere objects, the resulting magnetization varies for the same object in different orientations. (Theoretically, it is possible that in a certain orientation for some objects induced and permanent magnetization have the same strength, thus cancelling each other when opposed to each other. However, in most cases, permanent magnetization is much stronger than induced magnetization. Therefore, in practice there is always a magnetic signature that can be observed with magnetometers.) Also, because permanent magnetization varies from object to object, two objects of the same type will never exhibit the same magnetic field.

For survey planning, the case where an object is only subject to induced magnetization with no permanent magnetization and is oriented east-west with its longitudinal axis, can be assumed the worst case that must be detected. However, the north-south orientation for the purely induced case may be used as a compromise to define the survey objective. The induced magnetic field of UXO can be simulated as shown below in figure 2. For the below simulations, the objects are assumed to be cylinders with the diameter and the length of the UXO item in question. Inducing field values have been taken into account for an offshore wind farm in the Baltic in May 2014. Magnetic permeability is assumed with values for normal, unalloyed steel.

Generally, the total field magnetic amplitude observed over ferrous objects falls off with the third power of distance. This can be translated into the rule of thumb that the signal is reduced to 1/8 of the original values when the distance between the magnetometer and the object is doubled. For example, if the magnetic amplitude observed over a 250 lbs unexploded bomb is around 800 nT/m at a distance of 1 m distance between the magnetometer and the object, it will only be 100 nT/m at 2 m distance and 12.5 nT/m at 4 m distance.
Below, the magnetic signature expected for a 100 lbs bomb with purely induced magnetization is shown in figure 1. At 5 m below the total field magnetometer, the amplitude observed for the GP100 (100 lbs) bomb is only about 2.8 nT. In comparison, is still around 8 nT would be observed for the GP250 (250 lbs) bomb. Overall, the magnetic amplitudes indicated in table 1 can be expected as minimum amplitudes for objects which are subject to induced magnetization only.

![Figure 1: Minimum magnetic model for a 100 lbs unexploded bomb with purely induced magnetization oriented horizontally in east-west orientation at a survey altitude of 5 m. Grid 2 m, isolines 0.5 nT.](image)

<table>
<thead>
<tr>
<th>Magnetometer Altitude above Object [m]</th>
<th>Shell 105mm</th>
<th>Shell 155mm</th>
<th>Bomb 100lbs</th>
<th>Bomb 250lbs</th>
<th>Bomb 500lbs</th>
<th>G.Mine MKIV</th>
</tr>
</thead>
<tbody>
<tr>
<td>2</td>
<td>11 nT</td>
<td>41 nT</td>
<td>75 nT</td>
<td>160 nT</td>
<td>300 nT</td>
<td>1,350 nT</td>
</tr>
<tr>
<td>3</td>
<td>3 nT</td>
<td>12 nT</td>
<td>23 nT</td>
<td>49 nT</td>
<td>91 nT</td>
<td>485 nT</td>
</tr>
<tr>
<td>4</td>
<td>&lt;2 nT</td>
<td>5 nT</td>
<td>10 nT</td>
<td>21 nT</td>
<td>40 nT</td>
<td>223 nT</td>
</tr>
<tr>
<td>5</td>
<td>&lt;1 nT</td>
<td>2.5 nT</td>
<td>5 nT</td>
<td>10.5 nT</td>
<td>20 nT</td>
<td>120 nT</td>
</tr>
<tr>
<td>6</td>
<td>&lt;0.5 nT</td>
<td>&lt;2 nT</td>
<td>3 nT</td>
<td>6 nT</td>
<td>12 nT</td>
<td>70 nT</td>
</tr>
<tr>
<td>8</td>
<td>&lt;0.5 nT</td>
<td>&lt;1 nT</td>
<td>1 nT</td>
<td>2.5 nT</td>
<td>5 nT</td>
<td>30 nT</td>
</tr>
<tr>
<td>10</td>
<td>&lt;0.1 nT</td>
<td>&lt;0.5 nT</td>
<td>0.5 nT</td>
<td>1 nT</td>
<td>2.5 nT</td>
<td>16 nT</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Total Field Amplitude Observed for Object Oriented S-N with only Induced Magnetization</th>
</tr>
</thead>
<tbody>
<tr>
<td>Shell 105mm</td>
</tr>
<tr>
<td>11 nT</td>
</tr>
<tr>
<td>3 nT</td>
</tr>
<tr>
<td>&lt;2 nT</td>
</tr>
<tr>
<td>&lt;1 nT</td>
</tr>
<tr>
<td>&lt;0.5 nT</td>
</tr>
<tr>
<td>&lt;0.5 nT</td>
</tr>
<tr>
<td>&lt;0.1 nT</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Clearly detectable and distinguishable from geogenous objects</th>
</tr>
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<tbody>
<tr>
<td>Still detectable but not distinguishable from geogenous objects</td>
</tr>
<tr>
<td>Undetectable even in good offshore magnetometer surveys</td>
</tr>
</tbody>
</table>

Table 1: Expected minimum anomaly amplitudes for certain ferrous UXO.

When considering line spacing for offshore surveys it is necessary to analyze the spatial distribution of magnetic anomalies caused by ferrous objects. The 100 lbs bomb 5 m
below the magnetometer depicted in figure 1 exhibits values of only about 3 nT in a circular area with a radius of about 1.5 m around the positive maximum. The minimum exhibits values of less than 0.5 nT. With a total field anomaly of about 5 nT on the centerline over the object, the 100lbs bomb 5 m below the magnetometer is barely detectable but usually not discernible from geological background features which are often in the same amplitude range. At two meters offset from the centerline, the total field amplitude observed is in the range of 1 – 2 nT and thus in the typical noise range of offshore UXO surveys. This example clearly shows the need for low magnetometer altitudes (which can be derived from table 1 for different survey objectives) and small line spacings which are in the same range as must be expected based on common knowledge on land UXO surveys.

**METAL DETECTOR SURVEY**

Compared to magnetometers, metal detector systems have a reduced depth of detection. Also, 100% coverage must be obtained with metal detector systems and the coils be guided close to the seabed to guarantee detection of relevant objects. Therefore, metal detector systems are typically conducted with large (work class) ROVs at slow survey speeds (1 – 2 knots), resulting in high costs for this type of survey.

However, in certain cases where objects are expected buried in the sediments and magnetic background geology or noise / clutter prevent the application of magnetometers and acoustic systems, metal detector systems are the only alternative.

Generally, metal detector systems must be used whenever non-ferrous or demagnetized ferrous ERW / UXO objects such as German aluminium naval mines or demagnetized mines may be encountered.

**ACOUSTIC SURVEY**

Acoustic survey systems, namely sidescan sonar and multibeam echosounder systems should always be applied as auxiliary sensor systems in offshore ERW / UXO surveys. Even in cases where buried munitions are expected, objects clearly identified as harmless scrap in acoustic imaging data may be ruled out as ERW / UXO. However, it must be kept in mind that such objects may mask underlying UXO. Therefore this approach should be avoided in know munitions dumping areas.

In certain cases, e.g. where aluminium mines or demagnetized mines are expected on rocky / gravel seabeds, acoustic systems may be sufficient as primary sensors, in particular when considering survey efficiency compared to metal detector systems which would be the alternative.

Generally, high spatial resolution acoustic data is required (application of 900 kHz sidescan sonar systems at low survey altitudes) to allow for the differentiation of harmless scrap objects (barrels, pipes, etc.) from similar ERW / UXO items (small bombs, shells, etc.).

**COMBINED DATA INTERPRETATION**

Generally, more than one sensor should be used for ERW / UXO surveys to allow for combined data interpretation which will enable the interpreter to distinguish UXO from non-UXO in many cases.
3. SUMMARY

Offshore UXO detection required careful consideration of survey objectives, survey site parameters (in particular background geology) and survey system capabilities. Typically, magnetometers are the main sensors for the detection of buried ERW and UXO which largely have ferrous (steel) casings. However, acoustic systems are important as alternatives in cases where magnetometers and/or metal detectors are not applicable and as auxiliary sensors allowing to acquire more information on objects than on single magnetometer or metal detector datasets.

ERW and UXO surveys require high-resolution magnetometer, metal detector, sidescan sonar and multibeam echosounder data. Magnetometer altitude should be 3 m on average and not exceed 5 m while line spacing can vary between 1.5 and max. 5 m depending on the survey objectives. Metal detector surveys require 100% survey area coverage, low altitudes above seabed (0.5 – 1.0 m) and slow survey speed, making offshore metal detector surveys highly expensive. However, metal detector systems are the only alternative to detect ferrous and non-ferrous buried objects in geological backgrounds with strong magnetic effects.

Desirable, but not yet available is an acoustic (subbottom imaging) system that is capable of reliably detecting buried unexploded ordnance.

REFERENCES


OFFSHORE UNEXPLODED ORDNANCE RECOVERY AND DISPOSAL

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Abstract: Large quantities of excess munitions, including both conventional and chemical munitions, have been dumped in maritime environments worldwide. Also, military exercises, testing and warfare at sea have left behind large quantities of unexploded ordnance (UXO) at sea. Prior to offshore construction activities including geotechnical investigations, grapnel runs, trenching, pipe and cable laying, construction or ramming of foundations, risks from UXO must be excluded. During the UXO survey, typically many objects of interest are detected. Typically (except for dumping areas), most of the objects detected turn out to be innocuous during visual inspection. As operating costs are high for offshore UXO recovery and disposal, efficient technologies need to be applied for the task. Because of the water depths, environmental conditions (currents, visibility, and temperature) and hazards of UXO inspection and recovery, the application of technical, remotely operated systems must be preferred over divers. Issues that need to be addressed in offshore UXO recovery and disposal included

- Suitable high-precision underwater positioning for relocating of targets detected in geophysical surveys
- Sensor systems to assist the re-location of targets
- Maximization of operational time (24-hour-operations) for maximum efficiency
- Application of remotely operated vehicles with powerful manipulators and visual sensors (workclass ROVs) instead of humans (divers) for work on potentially lethal objects in order to reduce exposition and maximize productivity
• Application of tools for the recovery of bulk or heavy UXO or scrap (electromagnets, underwater excavators)
• Enhancement of underwater visibility to allow for the clear visual identification of targets (imaging sonars)
• Remotely operated placement of explosive devices and environmental protection considerations for open detonation of UXO at sea in case recovery and disposal on land are not possible

**Keywords:** Unexploded Ordnance, UXO, Investigation, Recovery, Disposal, Offshore, Underwater Positioning,
1. INTRODUCTION – UNEXPLODED ORDNANCE IN OFFSHORE ENVIRONMENTS

Unexploded ordnance (UXO) and dumped munitions pose considerable hazards to the marine environment in general and offshore construction activities in particular. Hazards include the release of toxic chemicals such as nitroaromatics, arsenic, white phosphorus and blister agents.

Offshore environments have been used excessively for dumping of conventional and chemical munitions in the past. Large quantities of munitions were dumped after World War I and World War II. In the post-World War II area, the coastal and high seas were used for the disposal of excess stockpile munitions of all kinds. Mine laying (figure 1) and marine warfare during the two World Wars (figure 2) left large areas of coastal waters contaminated with UXO. Military training and exercises in the inter- and post-war eras added further unexploded ordnance to the already high contamination.

For example, approximately 1.6 million metric tons of unexploded ordnance were dumped in the North Sea and the Baltic alone after World War I and World War II. In addition, some 215,000 – 235,000 tons of chemical munitions were dumped in the North Sea and the Baltic. During the two World Wars, approximately 600,000 mines were laid in the North Sea, and about 100,000 in the Baltic [1, 2, 3].

Munitions that pose risks to offshore construction such as pipe and cable laying or founding of wind turbines include large-calibre artillery shells, aircraft bombs, depth charges, mooring and ground mines, torpedoes and other ordnance containing large quantities of explosives because they may be initiated by activating their fuse mechanisms or simply by strong impact during construction activities. In addition, chemical warfare agents in bulk containers or filled in munitions pose hazards when brought to the surface where direct contact may occur.
Unexploded ordnance is a real hazard in offshore construction projects as the example of two wind farms in the German Bight illustrated in figure 3 shows. Within the project area of the two wind farms, nine naval mines and one torpedo, each containing more than 200 kg of high explosives were found and had to be detonated in situ because they were deemed unsafe to handle. In addition, several 10.5 cm to 28 cm calibre artillery shells were found at the site.

Figure 3: Example of UXO finds in the project area of two wind farms in the German Bight.

The above example clearly shows the need for UXO detection and removal operations prior to the realization of offshore construction projects.

2. OFFSHORE UXO RECOVERY AND DISPOSAL

Unexploded ordnance occurs in marine environments with water depths ranging from zero (tidal mudflats) to several thousand meters. However, current recovery and disposal projects are focused on continental shelves in water depths of up to approximately 100 meters where most of the offshore construction (cables, pipelines, wind farms, harbour and shipping channels) takes place.

Typically, UXO detection and removal is typically conducted in three phases. In the first phase, a desk study is conducted to assess the UXO risks for the project area. During the desk study, potential sources of UXO are identified. For this purpose, recent and
historic sea charts, literature, and archive documents are used. If a UXO risk is identified, a survey is designed and realized in the second phase. Typically, magnetometers, multibeam echosounder imaging and sidescan sonar imaging systems are used for offshore UXO surveys. Sometimes, additional sensors such as pulse induction metal detectors are applied. During the interpretation of the survey data, potential UXO objects are identified and marked for in situ investigation, recovery and – if necessary – in situ disposal by detonation. The latter are realized in the third phase of offshore UXO projects.

The third phase of any offshore UXO project requires suitable tools to

- relocate a detected object or its location on the seabed,
- confirm the existence of the object with a sensor,
- remove any sediments or other materials covering the object,
- visually inspect and identify the object,
- if necessary, enhance visibility in poor visibility (turbidity),
- recover the object (if possible and deemed safe to handle by explosive ordnance disposal specialists),
- if necessary, dismantle the object if it is too large for recovery in one piece,
- if necessary, position explosive charges to initiate in situ detonation of identified UXO that are classified as unsafe to handle,
- investigate the site for successful disposal and removal of any large debris and residual explosives after an in situ detonation.

During all offshore UXO recovery and disposal operations, the safety of personnel and the vessel is paramount. Therefore, unmanned, remotely operated and autonomous systems must be preferred over divers to reduce exposure to risks connected with the investigation, recovery and in-situ detonation of UXO.

**Remotely Operated Vehicles (ROVs)**

Remotely operated vehicles are the most suitable vehicles and tool platforms for offshore unexploded ordnance investigation and recovery. ROVs can typically be used in water depths of 10 meters and more. No suitable ROVs are available for shallower water at present, making it necessary to use divers in shallower areas.

ROVs have several advantages over divers, namely elimination of exposure of staff to unexploded ordnance, longer endurance, and higher weather criteria, as well as more versatile tools (acoustic sensors and imaging systems, cameras and lights, metal detector, magnetometers, manipulators) and higher lifting capacity.

While in some cases inspection class ROVs may be sufficient for simple camera inspections in calm waters, workclass ROVs have been found to be more reliable, robust platforms for offshore UXO inspection, recovery and removal operations because of their higher weather criteria (in particular waves, current), and more versatile usability (multiple tools and sensors)

**Underwater Positioning**

One of the most important factors for efficient and reliable relocation is highly precise underwater positioning. Typically, underwater acoustic positioning systems such as Ultra Short Baseline (USBL) systems in combination with inertial navigation systems (INS) are used for underwater positioning of ROVs and other equipment.
Considering the high costs of employing dynamic positioning (DP) offshore supply or dive support vessels with a workclass ROV, high position accuracy is important to enable quick relocation of targets. Typically, one meter positional accuracy that can be achieved with advanced USBL underwater positioning systems at 500 m range between the transceiver and the transponder are sufficient for the purpose of relocation.

**Simple Sensors**

Cameras and lights are suitable for visual inspection of the seabed and objects on the seabed or uncovered. However, they are limited to good visibility in relatively clear waters. While a diver can also rely on tactile sense, this disadvantage of the ROV is made up for by the possibility to have images of objects in question reviewed by several qualified members of staff (EOD specialists) and the fact that time is not a limiting factor for decision making when using an ROV as it is for divers who have limited time in the water.

**Detectors**

Detectors are necessary to confirm the relocation of an anomaly that was classified as potential UXO based on magnetometer, metal detector or sonar imaging data. Ideally, the same sensor is used for relocation as was used for the detection. Confirmation is particularly important if an object is buried below the seabed or located in turbid waters and cannot be immediately identified visually. The detector which may be a magnetometer array, metal detector array or imaging sonar is used to confirm the target based on a comparison with survey results.

In some cases, the investigation with the detector mounted on the ROV may be sufficient to exclude an object from further investigation, for example in cases where the metal detector shows no indication for an object that was detected in sonar imaging data or if an object that was suspected to be a large object is confirmed to be a small object of irrelevant size based on the combination of visual and metal detector findings during the relocation.

The detectors are also used to check the site of an object after the object has been removed to exclude underlying objects that are often found at munitions dump sites.

**Uncovering and Manipulation Tools**

Powerful dredge pumps and manipulators on workclass ROVs are suitable tools to uncover, move objects under investigation. While dredge pumps are primarily used to uncover objects buried in sediments (figure 4), manipulators are used to move and remove objects.

*Figure 4: ROV manipulator uncovering object with dredge pump.*
In cases where it is necessary to detonate unexploded ordnance in situ because the object in question is classified as unsafe to handle, the manipulators are used to put the explosive demolition charge in place.

Furthermore, additional tools for cutting, drilling or grinding may be adapted to an ROV to dismantle large objects prior to recovery. Usually, however, the ROV can be used to hook large items up to the crane of the accompanying DP vessel for recovery.

**Sonar Imaging Systems**

In high turbidity conditions that are often observed in tidal areas and seas in the continental shelves, visual inspection with cameras – even in combination with high power lights – is often limited by turbidity caused by sediments and algal bloom. In these cases, sonar imaging systems have been proven valuable tools (figures 5 and 6).

**Figure 5: Three 28 cm naval artillery shells visualised with an imaging sonar system in NIL visibility conditions in the North Sea.**

**Figure 6: Artillery shell dummies (cal. 7.5 - 15 cm) visualised with an imaging sonar system in NIL visibility in the North Sea.**

**Recovery Tools**

Depending on the task required, the ROV may use its manipulator to place items into baskets that are hoisted to deck by the crane of the ROV supporting DP vessel. In cases, where many small items (e.g. scattered small calibre munitions) have been identified and must be recovered, electromagnets with hydraulic flush systems may be used for recovery.

Only items that are classified as safe to handle are brought on board. After re-packing in special storage boxes, UXO recovered are typically stored in so-called wet storage areas (designated areas in the project area) and only recovered to be brought ashore by a dedicated transfer vessel. Thus, risk posed by the storage of unexploded ordnance on board are reduced to the absolute necessary minimum.
Preparing In-Situ Detonation and Post Destruction Inspection

ROVs are also used to position demolition charges onto unexploded ordnance that is classified as unsafe to handle or transport. Demolition charges are typically shaped charges. For underwater disposal operations, typically redundant charges are used. For the detonation, the ROV is typically recovered or positioned at a safe distance from the object.

Following an in-situ detonation, the tools of the ROV described above are used to inspect the detonation site to ensure the success of the operation, and remove any remaining debris and bulk explosives,

3. SUMMARY

Offshore UXO removal and disposal operations are ideally conducted using workclass ROVs. Acoustic systems are important components of the ROVs employed, in particular for precise underwater positioning during the relocation of objects and positions and enhancing or enabling visual identification of objects in low visibility / high turbidity conditions.

The accuracy of underwater positioning systems is in the range that is necessary for underwater UXO relocation. However, the stability of underwater positioning systems could still be increased. This would help to reduce downtime or extended relocations runs, in particular in rough weather conditions and situations where many units employing underwater positioning systems are working in the same project area.

Despite partial successes with sonar imaging systems to enhance visibility or enable visual identification of objects in high turbidity conditions, more systems for sonar imaging are desirable.

The development of acoustic initiator systems for underwater blasting operations (in-situ detonations) would provide an interesting alternative to cable-bound, shock tube or detonating fuse that is often difficult to handle in offshore environments with strong currents.

REFERENCES

USING A 3D SUBBOTTOM PROFILER FOR AUV-BASED PIPELINE DETECTION AND LOCALIZATION

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Abstract: A Deep Diving AUV (DD-AUV) with sensors and processing capacity to detect and to inspect surface laid and buried pipes/cable is presented. The detection and localization capability is achieved by fusing data from different sensors: optical cameras; magnetic gradiometers; side scan sonar; multi beam sonar and synthetic aperture sub-bottom profiler. The detection was performed on the basis of the Constant False Alarm Rate (CFAR) method.
This paper describes the current status of work on the vehicle; the sensors and the pipe/cable detection method.
Initial tests are performed with demonstrator testbed which verified the consistency of the hardware and software concepts.

Keywords: AUV, SBP, SAS, CFAR, pipe/cable tracking, buried object detection.
1. INTRODUCTION

ATLAS is building a robust, low cost, and reliable AUV demonstrator, capable of operating in great depths. This demonstrator is developed further in the Project KaPiTaS (Kabel and Pipeline Tracking System), which aims to develop an AUV-based pipe/cable inspection and localization capability. The implemented signal processing concept is based on fusion of magnetic, optical, and sonar based sensor systems. The detection and the localization algorithms are employed both for pipes/cable that lie on the seafloor and/or are buried within the seafloor. In this sense; the focus has been set to the use of the Buried Object Scanning Sonar [1] that uses a synthetic aperture algorithm to deliver a 3D representation of sub-bottom information. The application of this system has potential for detecting objects beyond the scope of the pipe/cable application [2], [3], [4], [5], [6]. The aim is to perform pipeline inspection with only one run over the pipeline using the detection and localisation algorithm to follow the pipeline autonomously with the AUV based on the merged data from its different sensors. This paper will give an overview of both: vehicle and the results obtained with the sub-bottom profiler.

2. SYSTEM DESCRIPTION

Initial tests have been performed on a testbed. During this time, a DD-AUV demonstrator is designed and built. This vehicle conveys a newly designed sensor suite aimed at giving a robust input for pipe/cable tracking algorithm. The following section describes the test bed, DD-AUV demonstrator and sensor suite.

2.1 Testbed description

Fig. 1 shows the test bed used for initial trials in order to minimize risk and dependency on the parallel development of the DD-AUV.

Fig. 1: Demonstrator testbed showing the basic layout of the KaPiTaS AUV: a flooded frame design, carrying sensors in a payload section in the nose and on a wing-like structure
The test bed vehicle has been modified to accommodate a wing-like structure, which houses the magnetometers, tracking cameras, and the receiver arrays for the 3D sub-bottom profiler.

2.2 Description of DD-AUV

The DD-AUV shown in Fig. 2 uses a system architecture taken from the M600-AUV, which since the beginning of the 2000’s has gradually evolved into a robust system with an excellent track record in terms of reliability. By combining this architecture with lightweight pressure tubes and low-complexity system components, a modular vehicle is produced. The AUV is easy to scale for various diving depths, while featuring ease of use, high endurance, and a short turn-around time on deck by exchangeable data storage and battery module.

Fig. 2: Basic concept of DD-AUV: open frame construction with easy access to all components

2.3 Overall sensor suite description

KaPiTaS started as Research and Development project with the aim of testing the applicability of various sensor systems for pipe and cable tracking. Therefore it features 5 different sensor systems:

1) Optical tracker (one HD camera for sequenced images, and two video cameras that provide continuous video stream), mounted at the front and at the wings of the AUV,
2) Magnetic tracker, mounted as a gradiometer with 8 3-axis sensors mounted on a wing-like sensor array,
3) Multi-Beam Echo-Sounder (MBES), mounted under the vehicle.
4) Side scan sonar (SSS), two arrays mounted at the sides of the vehicle; and
5) 3D Sub-Bottom Profiler (SBP), known as BOSS (Buried Object Scanning Sonar), the pinger is mounted under the vehicle (with a known offset), and 4 receiver arrays (each with 16 channels) mounted along the wings of the AUV.

This paper focuses on the detection results obtained with the SBP while surveying a fully buried pipeline.

3. TEST SCENARIO : LOCATING THE PIPELINE

A location was used, where two pipelines, running in parallel, can be found. They run stretch-wise on top of the sea floor and buried in the seafloor. As a first step, survey measurements were conducted using only SSS and MBES on a stretch where the pipes run on top of the sea floor. This served to locate the pipes and to establish a basis for estimating the pipes’ location in their buried stretch.

3.1 Preliminary Survey with MBES

The survey with the MBES clearly showed pipe structures in. An example is shown in Figure 3. The MBES operated at 400 kHz; at 8 m altitude and coverage swath of 40 m. The direction of the pipe is along the points indicated with R_1; R_2 and R_3 (in Fig. 3). A line was laid through these points and extrapolated towards the south west, where the pipes lie buried. Using the extrapolation, it was possible to roughly estimate the buried pipes location.

Fig. 3: 3D terrain View of the MBES data indicating the pipeline location
4. TEST SCENARIO: SEARCH FOR BURIED PIPE

In the second phase a survey was conducted only with the use of the SBP and the magnetometers. The expected geo location was estimated as a straight line along the previously detected non-buried pipeline.

4.1 Locating the area with the buried pipe

In the Fig. 4 a part of the SBP beamformed and SAS processed data is presented.

*Fig. 4: 3D Plan View of the SBP indicating the pipeline location*
The sonar window is generated with the BOSS system and the EdgeTech developed SAS processing (generated over 21 Pings). The horizontal and the vertical scale are in meters (indicated with yellow labels).

The sonar operates at a frequency of 5 kHz to 20 kHz (chirp generated signal) and a ping rate of 30 Hz. In order to assure good data quality (and overlapping of the footprints of subsequent pings) the selection of the ping rate depends on the AUV survey speed (~2 kn) and the height over ground (~3 m).

The central window shows the maximum backscatter intensity points generated at different sub-bottom layers (also named as Voxel Layers or VLs). In this case the beamforming was set to a sub bottom profiling from 0.5 m to 1.0 m relative to the sea floor value as detected at the DVL (Doppler Velocity Log). The seafloor is 0m and the Vertical positive direction is directed downwards (in the sediment). The horizontal (across and along track) and the vertical resolution are fixed to 5 cm.

The window at the right side is the profile as seen from left-to-right (maximum values through the VLs); and the window below is the profile as seen from top-to-bottom (maximum values through the VLs). The linear pipe structure is visible in the centre window as well as in the side windows (darker regions correspond to higher intensity backscatter).

The direction of the pipe is along the points indicated with P_1; P_2 and P_3. These points are on the same line as R_1; R_2; R_3 in Fig. 4.

It is worth noting than in this area there are two pipelines running in parallel. The second pipeline appears north from the main line at a distance of ~4.5 m.

4.2 Detection and localization of the pipe

The detection was carried out on the basis of the sonar data and by applying the CFAR method [7], [8], [9]. This method was adjusted to fit the calculation of high intensity backscatter caused by irregularities in the sediment profile. The parameters for the CFAR processing window were corresponding to the actual pipe size. At a resolution of 10 cm and pipe diameter of 80 cm the CFAR window size was set to 24 (the optimal search window is 3 times the pipe size over the resolution).

For the pipe detection and localization we use the detector on each VL; so the search is fixed at a single Voxel Depth. In this example the first VL is 0.5 m under the seafloor with 10 cm distance between layers; and up to 1 m under the seafloor. These settings resulted in 5 VLs.

As the detector calculates the target we can directly estimate the burial level; which equals to 0.5 m + (VL number -1)*0.1 m. In Fig. 5 a snapshot of evaluation windows is shown; the detections are calculated over 100 Pings.

On the left hand side the CFAR calculation values are shown (for a single ping): blue - sum over the cross single Voxel values (this is the input signal); green – mean value over the signal; cyan - median value over the signal; red – target detection. In Fig. 5 these values are from the last ping. On the right hand side there are two values: red (value of 1) for detected targets; and blue (value 0) for no-detection. These values are automatically correlated to the AUV’s position; heading and distance from the central line; thus allowing the localization of the target on and in the seafloor. The red lines in VL 1, VL2 and VL 3 are within the exact region that corresponds to the high level (pipe induced) backscatter within the points P_2 up to P_3 (as shown in Fig. 4).
Fig. 5: CFAR evaluation example over 100 pings; base 3D sub-bottom data as shown in Fig.4

3. CONCLUSIONS

A deep diving AUV is developed with ability to perform detection and localization of both: buried and non-buried pipes/cable. This capacity is being accomplished by using optical, magnetic and sonar sensors. The advantage of this setting is a higher detection due to the different levels of interaction between the pipe/cable and the sensors.

This paper also showed the preliminary results obtained with the MBES, the BOSS sonar and the implemented CFAR method.

Initial tests in real sea environment demonstrated robust performance of the hardware and the software concepts.
4. ACKNOWLEDGEMENTS

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Session 31

Unmanned Vehicles (AUV, USV and Gliders) for Underwater Acoustic Surveillance and Monitoring

Organizers: Alain Maguer, Brian Ferguson and Eric Delory
MONITORING THE UNDERWATER ACOUSTIC PRESSURE FIELD USING TWO SPATIALLY-SEPARATED HYDROPHONES WITH APPLICATION TO FORWARD-AFT SENSORS ONBOARD AN UNDERSEA GLIDER

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Abstract: A single hydrophone onboard an undersea glider has been used previously to detect and range surface watercraft for passive surveillance and collision avoidance purposes. The instantaneous range of a source can be estimated by measuring the multipath delay between the direct and indirect (boundary-reflected) path arrivals of the signal at the sensor. The addition of another sensor enables the estimation of the instantaneous source bearing by measuring the time delay between the direct path arrivals of the signal at the two sensors. To enhance the precision of the bearing estimates, the two sensors need to be mounted on the forward and aft extremities of the glider so as to maximize the intersensor separation distance. In this paper, the acoustic data recorded from two fixed hydrophones (positioned 1 m above the sea floor and 14 m apart) in a shallow water experiment are processed to obtain time delay measurements by either cross-correlating the outputs of the hydrophones for continuous signals or differencing the time of arrival of the signal at each of the hydrophones for transient signals. Each time delay measurement is then converted to a source bearing estimate. The variations with time of the instantaneous bearing estimates are shown for the transits of various continuous sound sources: an autonomous underwater vehicle (AUV), a diver propulsion vehicle, a helicopter, a maritime services work boat, a rigid-hulled inflatable boat, and an open-circuit scuba diver. Also, the time delay and source bearing estimates for a series of underwater acoustic communication transmissions from the AUV are presented. The time delay measurement errors of these transmissions match those previously observed for mechanical transients and biological sonar pulse transmissions. The bearing measurement error for each of the sources depends on the intersensor spacing, source bearing and time delay measurement error. For a typical undersea glider fitted with two hydrophones (one forward, the other aft), the bearing measurement error would be an order of magnitude higher due to the smaller intersensor separation distance.

Keywords: Time delay estimation, bearing estimation, passive sonar signal processing
1. INTRODUCTION

Traditionally, pervasive wide area surveillance of the undersea acoustic environment is achieved by deploying large arrays of hydrophones at a central location and using both temporal and spatial filtering to detect weak signals, resolve closely-spaced sources and estimate the bearing and other properties of a signal source. Optimal (adaptive) beamforming is implemented to maximize the array gain, minimize spatial leakage (sidelobes), and suppress (null) sources of interference [1]. Also, in practice, it is found that adaptive beamformers enable sensor arrays to be superdirective i.e., to provide array gain (or form narrow beams) at frequencies well below the design frequency [2]. This centralized approach is adopted, for example, by submarines, which are at one extreme (complexity) of the underwater vehicle spectrum. An alternative (decentralized) approach is to populate the area with (ideally, massively deployed) low cost, low capability sensor nodes that form an ad hoc sensor network. Rather than an array with hundreds of sensors and centralized processing, each node of the network often has just one sensor with basic signal processing to extract the tactical information, which then undergoes decentralized data fusion with the tactical information from neighbouring nodes, for subsequent relay via a gateway to inform the intelligence and decision making processes. The idea of sensing the underwater environment with just one hydrophone, either fixed [3], or onboard an undersea glider [4] has been considered previously by the authors.

Undersea gliders are at the other extreme (simplicity) of the underwater vehicle spectrum. For an acoustic sensor onboard an undersea glider, the effect of hydrodynamic flow noise (or broadband pressure fluctuations induced by unsteady flow or turbulence) is small and platform noise is observed only briefly when either the buoyancy change pump or trim adjustment mechanism is actuated. Also, it has been proposed that gliders act as mobile nodes in distributed autonomous sensor networks for surveillance of the undersea environment to deny quiet diesel-electric threat submarines unmonitored movement in focal areas [5]. This approach uses gliders in littoral waters as both sensing nodes and low-observable acoustic-satellite communication gateways to relay tactical data from other nodes (such as fixed bottom-mounted sensors) in the network. An autonomous undersea sensor network could be covertly deployed in days, operate for weeks to months, and adapt to in situ conditions to provide a capability for the detection, classification, localization, and tracking of subsurface threats.

Previously, acoustic sensor data from a single hydrophone onboard a Slocum glider were recorded during the glider’s deployment in a shallow water environment with a typical water depth of 90 m. Post processing of the data resulted in the detection of a helicopter, a surface vessel, tone burst transmissions from an underwater sound projector, and echolocation clicks emitted by Tursiops aduncus dolphins that frequent the area. It has been shown previously by the authors that tactical information such as speed, altitude and rotor/propeller blade rates of transiting air vehicles (helicopters and turboprop aircraft) can be obtained using a time-frequency method [6]. The spectrograms of the acoustic data recorded for two different time periods showed, respectively, the acoustic characteristics of a slowly moving helicopter and two different transmission frequencies for an underwater sound beacon operating in the area. A cepstrum analysis of the acoustic data recorded for another time period provided an estimate of the multipath delay between the direct path and multipath arrivals of a surface vessel’s radiated sound at the hydrophone as a function of time. These multipath delay estimates, along with the known (measured) water depth, hydrophone depth, and sound speed in water were input to a single hydrophone multipath passive ranging method [7].
variation with time of the resulting range estimates provided the distance and time of the closest point of approach of the vessel to the glider. Also, waveforms of sound pulses emitted by local dolphins were observed using the single hydrophone onboard the glider.

The present research aims to extend the capability of a single hydrophone (located aft) onboard an autonomous undersea glider to monitor the underwater acoustic pressure field by adding another hydrophone (located forward) on the glider. Section 2 of the paper considers the use of two spatially separated hydrophones to measure the time delay, i.e., difference in the times of arrival of the signal at the two hydrophones, and its transformation to a farfield bearing estimate. The time delay measurement errors encountered in practice are quantified, along with the bearing estimation errors for both transient underwater acoustic signals (AUV acoustic communication transmissions (ACOMMS), dolphin biosonar pulse transmissions, and hammer strikes) and continuous signals from surface vessels, underwater vehicles and a helicopter. Section 3 presents the observed variations with time of the source bearings during the transits of an AUV, an underwater scooter, a helicopter, and a work boat. The conference presentation will also include the observed variations with time of the source bearings during the transits of a RHIB [3] and an open-circuit scuba diver [8]. The conclusions are given in the final section.

2. SOURCE BEARING ESTIMATION USING TIME DELAY MEASUREMENTS

When the source is in the farfield, the wavefront arriving at a pair of spatially separated sensors is planar. The cosine of the source bearing \( \beta \) (measured with respect to the sensor pair axis) is given by

\[
\cos \beta = \frac{c \tau}{L},
\]

where \( L \) is the intersensor separation distance, \( c \) is the speed of sound propagation in water and \( \tau \) is the difference in the times of arrival of the signal at the two sensors, which is commonly referred to as the time delay. Let \( \varepsilon(\beta) \) and \( \varepsilon(\tau) \) denote the respective errors in bearing \( (\beta) \) and time delay \( (\tau) \). It follows from (1) that

\[
\varepsilon(\beta) = \frac{-c}{L \sin \beta} \varepsilon(\tau).
\]  

Experiments were conducted to measure the time delays for two classes of underwater acoustic signals: transient and continuous. The two hydrophones were 14 m apart. The transient signals consisted of AUV underwater ACOMMS, dolphin biosonar pulse transmissions [9], and mechanical transients (generated by hammer strikes) [7]. Figure 1 shows that for the transient signals, the standard deviations of the time delay measurements are \( \leq 1 \) \( \mu \)s. Figure 2 shows the bearing estimates of the AUV underwater ACOMMS for multiple vehicle transits during a 20 minute interval.

For the continuous sound sources (AUV, underwater scooter, helicopter, work boat, RHIB, and open-circuit scuba diver), the standard deviations of the observed time delays are \( \leq 100 \) \( \mu \)s. Table 1 provides a summary of the observed time delay errors and source bearing errors when the source bearing \( \beta = 60^\circ \). For the present experiment, where \( c=1520 \) m/s and \( L=14 \) m, \( \varepsilon(\tau) \leq 1 \) \( \mu \)s (\( |\varepsilon(\beta)| \leq 0.007^\circ \)) for transient signals and \( \varepsilon(\tau) \leq 100 \) \( \mu \)s (\( |\varepsilon(\beta)| \leq 0.7^\circ \)) for continuous signals. For an assumed forward-aft hydrophone separation distance \( L=2 \) m, the predicted bearing errors are \( |\varepsilon(\beta)| \leq 0.05^\circ \) for transient signals, and \( |\varepsilon(\beta)| \leq 5.0^\circ \) for continuous signals.
Fig. 1. Standard deviations of time delay measurements for acoustic transient signals.

Fig. 2. Variations with time of bearing estimates of AUV underwater ACOMMS for nine vehicle transits.

<table>
<thead>
<tr>
<th>SIGNAL CLASS</th>
<th>TIME DELAY ERROR</th>
<th>BEARING ERROR L = 14 m</th>
<th>BEARING ERROR L = 2 m</th>
</tr>
</thead>
<tbody>
<tr>
<td>Transient</td>
<td>≤ 1 µs</td>
<td>≤ 0.007°</td>
<td>≤ 0.05°</td>
</tr>
<tr>
<td>Continuous</td>
<td>≤ 100 µs</td>
<td>≤ 0.7°</td>
<td>≤ 5.0°</td>
</tr>
</tbody>
</table>

Table 1: Time delay and bearing estimation errors for a source bearing of 60° with intersensor spacing L=2 and 14 m.
3. CONTINUOUS UNDERWATER ACOUSTIC SIGNALS

Time delay estimation for a continuous underwater acoustic signal radiated by a source in motion involves taking the time series outputs of the two spatially separated hydrophones and computing the generalized (phase transform) cross-correlation function over short (subsecond) time intervals for the frequency band 20 – 1200 Hz; the estimate of the time delay is equal to the time lag at which the cross-correlation function attains its maximum value. Figure 3 shows the variation with time of the cross-correlation function during a transit of an underwater scooter in a direction parallel to the sensor pair axis. The time delay estimate for each subsecond interval is converted to a bearing and the results are plotted in Fig. 4. The transition of the scooter from one endfire direction to the other is evident.

![Figure 3. Variation with time of cross-correlation function for an underwater scooter transit.](image)

![Figure 4. Variation with time of source bearing estimate for an underwater scooter transit.](image)
Similarly, Figs 5 and 6 show (respectively) the variations with time of the cross-correlation functions (as functions of bearing rather than time delay) and the instantaneous source bearing estimates for nine transits of an AUV. The source bearing estimate at a given time corresponds to the bearing at which the cross-correlation function at that time attains its maximum value. Each AUV transit is in a direction parallel to the axis of the sensor pair.

**Fig. 5. Variations with time of cross-correlation functions for nine AUV transits.**

**Fig. 6. Variations with time of source bearing estimates for nine AUV transits.**
Finally, Figs 7 and 8 show the respective variations with time of the cross-correlation functions (as functions of bearing rather than time delay) for the transits of a helicopter and a work boat.

Fig. 7. Variation with time of cross-correlation function for a helicopter transit.

Fig. 8. Variation with time of cross-correlation function for a workboat transit.
5. CONCLUSIONS

The capability of a single hydrophone (located aft) onboard an autonomous undersea glider to monitor the underwater acoustic pressure field can be extended by the addition of another hydrophone (located forward) on the glider. The hydrophone pair would provide directional information on surface vessels, underwater vehicles, air vehicles, echolocating marine mammals, open circuit scuba divers, and underwater acoustic transient sources.

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PASSIVE ACOUSTICS EMBEDDED ON GLIDERS – WEATHER OBSERVATION THROUGH AMBIENT NOISE.

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Abstract: The ability to associate underwater glider data with estimates of surface weather conditions allows for a novel approach to study air-sea interactions. The development of a glider embedded weather sensor has been studied. Our approach was based on the WOTAN methodology. In the 1kHz-30kHz frequency range, the background underwater noise is dominated by wind generated noise. Focusing on the sound pressure level at 5kHz, 8kHz, 10kHz and 20kHz, we were able to provide an estimation of the sea surface wind speed. Thus, deploying a glider with an embedded hydrophone means we are now able to have direct access to the surface weather conditions around its position. To further investigate this capability, we deployed several gliders with passive acoustic monitoring devices onboard in the Mediterranean Sea. This was during the 2013 Moose and HyMeX/MERMeX experiments. Four months of data have been recorded and post recovery processed. Wind speed estimations have been compared to weather buoys observations and operational atmospheric models predictions. The wind speed estimates were obtained with an error of approximately \( \sim 2\text{m.s}^{-1} \). A specific emphasis has been placed on the robustness of the processing through multi frequencies analysis and depth induced attenuation correction, as well as on the acoustic sampling protocol on which a downscaling study has been performed to meet the low energy consumption glider standards, for a future real time embedded processing.

Keywords: Wotan, passive acoustics, wind speed, glider
INTRODUCTION

Underwater gliders can provide high resolution (~4 h / 4 kms) ocean temperature and salinity profiles. Being able to associate them with estimates of surface weather conditions would enable a better study of air-sea interactions. Since in-situ observations of the marine meteorological parameters are difficult, the development of a glider embedded weather sensor have been studied and was based on the WOTAN approach (Weather Observation Through Ambient Noise), as described by Nystuen et al. 2010 [1]. Wind generated noise is predominant in the underwater background noise in the 1kHz to 30kHz frequency band, as shown on the figure 1, and there’s a linear relation between wind speed and sound pressure level (spl).

We have deployed gliders in the Mediterranean Sea, with passive acoustic monitoring devices onboard during the 2013 Moose and HyMeX/MerMeX experiments. 4 months of data have been recorded, with recurrent opportunities to compare our estimations to the MeteoFrance weather buoys in the area.

Focusing on the spl at 5kHz, 8kHz, 10kHz and 20kHz, we can discard unknown noisy events from wind events judging by the spectrum’s slope and in doing so estimate the surface wind speed around the glider’s position (~10 km²).

DATA ACQUISITION

We used the Acousonde B003A-HF, a data logger developed by Greeneridge Sciences Inc. They specifically developed an external tethered battery pack for the Acousonde,
raising its estimated autonomy from 50 hours to 200 hours of continuous recording. The Acousonde is totally independent from the glider. It has its own battery, memory and programmed mission. We mounted it on Slocum gliders, as shown on fig.2 and the data processing was made after the glider has been recovered.

![Acousonde on glider Tintin. Photo by P. Cauchy](image)

**Fig.2: Acousonde on glider Tintin**

We deployed the gliders with the mounted Acousondes on Moose / HyMeX / MerMeX gliders, in the North Western Mediterranean basin. The gliders were piloted along transects that closely passed the Meteofrance weather buoys at Lion and Azur. Thus, allowing for recurrent calibration/validation of the measurements made.

We followed the data acquisition protocol described by Nystuen et al. 2010 [1], however, tailored to meet the conditions of our experiment. The Acousonde’s recording autonomy is 10 times lower than the usual glider mission time. To optimise the autonomy and time resolution, we decided to record 1 minute out of 10, sampled at 50kHz rate.

4 deployments have been performed, representing a total amount of 8 months of data, among which 7 weeks are in the weather buoys area (distance < 40 km.).

**DATA PROCESSING**

Data cleaning has been performed according to the Acousonde’s pressure sensor. Data deeper than 950m is removed, to avoid the glider’s pump noise (usual dives depth is 1000m). Data shallower than 20m has also been discarded to avoid the glider generated noise at the surface (pump, communication).

On each 1 minute recorded sample, we compute an average spectrum as follows:

20 spectra are computed on 204.8ms. time series, each separated by 0.5s. Each time series is fast Fourier transformed (FFT) to obtain a 512-point (0-50kHz) power spectrum. Each spectrum is integrated on third octave frequency bands. An average spectrum on these 20 spectra is computed, representing 10s of signal. This is repeated 4 times, every 15s to represent the whole 1min sample. A stationarity criterion is applied to the 4 resulting spectra, to discard transient sounds. Each of the 4 spectra is compared to the 3 other’s mean (at 5, 8, 10 and 20kHz). Each spectrum that differs from the others by more than 10% is discarded. If several spectra are discarded, the whole sample is discarded. The average spectrum is then computed.

Samples with a positive spectrum’s slope, judging by spl at 5, 8, 10 and 20kHz, are classified as non-wind generated and discarded. The spl at 8kHz time series is then smoothed, using a 3 hours window moving average, and used to estimate the wind speed.
One glider deployment (MooseT00_25) is kept apart, as an evaluation dataset. We consider the 3 other deployments as a training dataset (MistralT02_01, ASICSMED and MooseT00_23). They represent a supervised dataset of 1442 sound pressure level measurements that are in a weather buoy’s area (i.e. distance to the buoy < 40 km), in a 14 weeks overall dataset.

To estimate the parameters, we performed a linear regression on a homogeneous dataset, considering only 40 random measurement points in each 5m.s\(^{-1}\) wind speed band. The operation has been repeated 100 times for robustness. The linear regression between spl(8kHz) and the buoy measured wind speed, as well as the related error distribution, are shown in figure 3.

![Fig.3: Linear regression results: slope = 7.45 (m.s\(^{-1}\) dB\(^{-1}\)), offset = -460 (m.s\(^{-1}\)) Error’s standard deviation = 2.7 (m.s\(^{-1}\))](image)

The wind speed estimates of the training dataset are shown in figure 4. Anemometer’s measurement is the blue thin line, glider embedded acoustic based estimation is the red thick line, and the distance from the glider to the buoy is the dark interrupted line.

![Fig.4: Wind speed measured by the buoy’s anemometer (thin blue line) Glider’s estimation (red thick line), and distance to the buoy (dark interrupted line)](image)
RESULTS

1.1. Evaluation dataset

We used the parameters from the learning dataset linear regression to estimate the wind speed of the evaluation deployment. The wind speed measurements are shown in the figure 5. Comparison with the anemometer’s measurement when the glider is close to the buoy shows a -2.1 m.s\(^{-1}\) mean error and 1.7 m.s\(^{-1}\) standard deviation.

![Fig.5: Wind speed measured by the buoy’s anemometer (thin blue line) Glider’s estimation (red thick line), and distance to the buoy (dark interrupted line)](image)

1.2. Multi-frequencies analysis

Considering the regularity of the wind generated noise’s spectrum, in the 1 kHz – 30kHz frequency range, we considered widening the frequency band studied, to remove some noise. We have repeated the study, focusing on the third octave centred on 10kHz. Results in wind speed estimation are similar to the previous results, as well as the linear dependency parameters. Considering the mean wind speed, between the 8kHz and the 10kHz estimation, the results are slightly improved, as shown on the figure 6.

![Fig.6: Wind speed estimation error distributions on the training dataset Wind speed estimated à 8 and 10kHz, and 8 – 10kHz mean.](image)
1.3. Depth induced error

In this study, we used a 3 hourly moving average smoothing. Since it’s close to the glider’s dive duration, it attenuates any depth related effect on wind speed measurement. Without this smoothing, the wind speed estimation shows oscillations that are obviously related to the gliders successive dives. The spectra of the glider’s depth (dark interrupted line) and sound level time series (red line) are plotted on the left panel of the figure 7. To estimate the depth related error, we compared the sound pressure level to its mean on a 5 hours window. The right panel of the figure 7 shows the time series of the depth related error, the colour represents the depth. The 3 training deployments are plotted successively. The spectrum study shows that there is a depth related effect. The time series suggests that the effect is not constant, and the error induced is less than 0.5 m.s\(^{-1}\).

![Fig.7: Right: Depth related error time series. Left: Spectrum comparison. Sound level (red line), depth (dark interrupted line)](image)

1.4. Wind direction

When at the surface, a Slocum glider is affected by surface conditions (waves, wind). Empirically, we observed that the glider heads toward the wind direction, which can be explained by a vane effect, induced by its tail up in the air. The figure 8 shows the gliders heading evolutions during a deployment, the lower panel being a zoom on a few dives. It appears that the gliders surface heading is coherent during several dives, and is independent from its underwater heading.

We have performed a very short glider test deployment, close to the shore. We extracted the surface heading from the glider data, and compared it to the wind direction given by the St Mandrier and Toulon weather stations. The results are shown in figure 9.
1.5. Towards a glider integrated system

The feasibility of a glider embedded WOTAN system has been explored in this study. The glider generated noise and its vertical movement are not perturbing the estimation. Moreover, the surface behaviour of the Slocum glider allows an estimation of the wind direction.

Wind speed has been estimated with an error standard deviation of $\sim 2\text{m.s}^{-1}$, which could be improved through depth effect correction. The results observed on the evaluation deployment show an underestimation, which could only be corrected by a cross validation event during the deployment (e.g. the glider being close to an anemometer). A specific emphasis should be placed on the acoustic sensor, whose absolute value reliability could be improved by a prior calibration.

The challenging part, in the development of a glider integrated WOTAN sensor will be the data acquisition energy consumption. We have studied a few data acquisition protocols, that may lower the energy consumption, but a specific hardware development, towards a low power sensor and acquisition system is mandatory.
REFERENCES

SMALL VESSEL DETECTION THROUGH THE USE OF AN UNDERWATER GLIDER

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Abstract: Monitoring the marine traffic of small and mid-sized boats is of major interest for many applications, including surveillance against illegal trafficking of goods and illegal immigration. The approach presented in this article is conducting passive acoustic monitoring of marine traffic by hosting a small volumetric hydrophone array of eight elements on an underwater glider. Passive underwater acoustic monitoring technologies applied to mobile autonomous underwater platforms may allow: minimum environmental impact, covertness, long endurance, wide area coverage, near real-time, continuous (‘24/7’) monitoring, availability of several functionalities, ranging from detection to classification of multiple acoustic noise sources at the same time. This paper describes the mobile acoustic measurement system, presents the work in progress in terms of the processing chain developed for detection and estimation of the direction of arrival with respect to the sensing platform. The experimental results obtained from data collected at-sea with a similar antenna during preliminary at-sea tests are discussed. This work is partially funded by the European Union in the context of the FP7 PERSEUS Project.

Keywords: Marine traffic acoustic monitoring, small boat detection, underwater gliders
1. INTRODUCTION

Monitoring the marine traffic of small and mid-sized boats is of major interest for many applications, ranging from the surveillance against illegal immigration to the protection of marine parks or assets. In this work, which is performed in the context of the FP7 European Research Project PERSEUS [1], the approach selected is conducting passive acoustic monitoring of marine traffic by hosting a small volumetric hydrophone array on an underwater glider.

The Centre for Maritime Research and Experimentation (CMRE), which is part of NATO’s Science and Technology Organisation (STO), has extensive experience operating gliders: CMRE owns a glider fleet consisting of six shallow (<200m) and two deep (<1000m) Slocum gliders from Teledyne Webb Research. The Slocum Glider is an underwater autonomous vehicle driven along saw-tooth vertical paths by variable buoyancy. Able to host a wide variety of sensors, it can be programmed to patrol a sea area for long time, periodically surfacing to transmit its data to shore while downloading new instructions. CMRE gliders can be equipped with various oceanographic, optical and acoustic payloads. If gliders have been used in the past mainly for large-scale oceanographic and environmental surveys, in recent years the glider technology and their application to underwater sensing have matured, allowing its capabilities to be expanded to include underwater passive acoustic sensing operations. Passive underwater acoustic measuring technologies applied to such mobile autonomous underwater platforms show a variety of advantages, among which are minimum environmental impact and covertness, long endurance, relatively easy deployment and recovery, on-board processing capability and autonomous robotic behaviour.

In this work, the addressed objective is to detect and estimate the direction of arrival of boats by processing acoustic data directly on board the glider through the application of appropriate algorithms. The processing chain will be applied to the acoustic data on the vehicle during its dive and the results will be sent to the glider’s control station through ‘instant messaging’ via IRIDIUM satellite connection each time it comes to the surface.

This paper will describe the mobile acoustic measurement system, present the processing chain developed, and show the experimental results obtained by applying the proposed methodology to an existing acoustic measurement system.

2. THE GLIDER ACOUSTIC PAYLOAD

An acoustic aperture in the form of a compact volumetric (3D) array is installed on the nose of the glider, augmented with a hydrophone spatially separated from it in order to have longer-baseline measurements available and, thus, localize targets. All hydrophones’ data are simultaneously sampled at high frequency (between 100 and 140 kHz) to exploit the wide acoustic bandwidth of the signals of interest and to have high resolution cross-correlation data available from pairs of hydrophones. In addition to the acoustic payload, the glider is equipped with a CTD (multi-parameter oceanographic sensor), which is able to feed the software tools of signal processing with measurements of the local sound speed profile and of the vehicle’s depth in real time; also, a motion reference unit has been installed to provide the instantaneous orientation of the vehicle. When the glider is at the sea surface its geographical position is available as well through its GPS antenna.
3. THE PROPOSED PROCESSING CHAIN FOR DETECTION AND DIRECTION OF ARRIVAL ESTIMATION

Appropriate algorithms are developed to automatically detect surface vehicles (fast boats in particular) from the volumetric array of hydrophones hosted on the glider. The processing chain will be eventually applied to acoustic data on board the glider during its dive. Suitable array processing algorithms are applied to the conditioned acoustic signals in order to determine the Direction Of Arrival (DOA) of fast boats passing in the area and possibly (depending on the geometry and the in-situ environmental conditions) provide additional outputs such as the vessel class (typically three: small, medium and large vessels) and the target course and speed. The paper addresses the description of the methodological approach for the array processing, detection and DOA estimation tasks.

The selection of proper array processing approaches for source separation and DOA estimation is driven by the correlation between the nature of the signal of interest (in our case, continuous, broadband, non-stationary) and the geometry of the hydrophone array. The bandwidth of noise radiated by small fast boats generally ranges from a few Hz to tens of kHz [2]. The geometry of a volumetric array which has to fit on the nose of a glider is characterized by a relatively small physical aperture and relatively low power consumption. For this reason the resolution of the array beampattern is necessarily limited. The proposed approach is to exploit the signal full bandwidth by fusing the results of a 3D beamformer, which is applied to the lower frequency sub-band of the signals sensed by the array, with the results of a TDOA approach, which is applied to the higher-frequency sub-band (see Fig. 2).

![Fig. 1: The CMRE Slocum glider equipped with the GLASS array plus a rear hydrophone.](image)

**Fig. 2: Block diagram of the array processing and DOA estimation methods.**
The concept is preliminarily proven by using an existing acoustic array of 8 elements, called GLASS array (Fig. 1). It is characterized by a regular small tetrahedron (hydrophone spacing = 0.1 m) plus a 0.4 m vertical array of 5 elements (one in common with the tetrahedron), again equally spaced by 0.1 m. Hence, overall, it can be seen as the combination of a 3D uniform array (with a spacing such that half-wavelength constraint of plane-wave beamforming is satisfied for $f \leq 7.5$ kHz), and a linear array with longer aperture (4 times more than the other), which privileges DOA estimate on the vertical plane in the same bandwidth as the tetrahedron.

### 3.1 MVDR adaptive beamformer

An adaptive MVDR beamformer [3] is applied to the whole 8 element array in the bandwidth 0.01-12 kHz (where the upper bandwidth bound overcomes the abovementioned half-wavelength constraint by exploiting the broadband nature of the signal to keep the grating lobes low). In the presence of multiple targets, working on the beamformed image is very convenient as it directly provides the azimuth and elevation estimate of each detected target, although, due to beam fatness, its ability to resolve close targets is limited (see examples in Fig.s 3 and 4).

#### 3.1.1 Normalization approach for false alarm reduction

A normalization approach is applied in order to decrease the number of possible false alarms deriving from side lobe and grating lobe effects. This method is more effective on longer apertures, i.e., when the angular resolution is higher.

The method starts with the $N$ highest local maxima from the raw beamformed image for the azimuth and elevation angles considered. It is clear that for a compact array, the normalizer will work reliably only for a limited number of targets. Starting from the highest local maximum, the two-directional normalizer will be applied. For each maximum, the main lobe will be excluded from the noise estimation process (with a size of $[\text{constant} \times 2\Theta_{\text{azimuth}}]$ times $[\text{constant} \times 2\Theta_{\text{elevation}}]$), as well as the predicted side- and grating lobes (cf. Figs. 3 and 4). Also, the interference through the sidelobes of the other maxima will need to be excluded. The above exclusion of ‘interference regions’ effectively creates a mask that will need to ensure the appropriate measurement of the background noise statistics. Once this normalization mask is applied, an energy detector can be used, the threshold of which is determined on the basis of the statistical analysis of background noise (Fig. 5). In very shallow waters (i.e., if water depth is in the order of few tens of meters), the elevation estimate accuracy achievable by the beamformer is not sufficient to provide a good range (hence 3D localization) estimate. In order to improve particularly the elevation estimate accuracy, a method of direction of arrival estimation is applied which is based on signal time coherence between hydrophone pairs (TDOA).

### 3.2 TDOA approach

The TDOA approach [2][4] is based on:

a) A generalized cross-correlation estimate [5] for each hydrophone pair $ij$, from which to compute the difference of time of arrival $\tau_{ij}$ of the signal on each pair hydrophone,

b) The Least Squares Method (LSM) to solve the equation $\tau = k^T d$, where $\tau$ is the vector of time differences measured from each pair of selected hydrophones, $k$ the source wave vector to estimate, from which azimuth and elevation can be easily derived [2], $T$
indicates vector transpose, and \( \mathbf{d} \) the vector of the difference of the 3D positions of each pair of hydrophones.

**Fig. 3:** Theoretical beam pattern of GLASS array when a plane wave comes from the direction \( \theta=0^\circ, \phi=0^\circ \).

**Fig. 4:** Theoretical beam pattern when a plane wave comes from direction \( \theta=45^\circ, \phi=20^\circ \).

**Fig. 5:** Sketch of normalization masks applied to a beamformed image if two targets 1 and 2 (red and blue respectively) are detected. Masked sidelobes/grating lobes are labelled with the same number as the related target. Over the grey regions statistical analysis of background noise can be conducted.
This approach exploits the broadband nature of the signal of interest, and needs a data sampling rate much higher than the selected maximum frequency to get high resolution in the cross-correlation function. TDOA is applied to a sub-array consisting of the most distant hydrophones of the volumetric array; it may provide an accurate estimate of the wave-vector $k$ of a noise source, but it is hard to use when multiple sources are present at the same time. For this reason, the idea proposed is to feed the TDOA approach with the rough estimate of azimuth and elevation angles provided by the beamformer for each detected target, as a guideline for separating the various noise sources, and then to refine each estimate by the TDOA approach.

In the case of the GLASS array, this approach is applied only to the extreme hydrophones of the vertical sub-array (spacing = 0.4 m). The spacing is enough to apply the algorithm to signals in the bandwidth beyond 3 kHz (again exploiting the ability of broadband signals to reduce grating lobes). The signal is also low-pass filtered at 40 kHz as its level is supposed to decrease rapidly at higher frequency for distances larger than a few hundred meters. Using only a vertical pair implies the possibility to estimate only the target elevation. Hence, in the present, a simplified version of the methodology for the elevation estimate provided by the TDOA approach is used along with the azimuth estimate from the beamformer in order to get a 3D localization.

4. PRELIMINARY EXPERIMENT AND RESULTS (PROOF OF CONCEPT)

A selection of results are presented which were obtained by the application of the proposed method to the acoustic data recorded by the 8-element array during GLASS’12 sea trial in front of Marina di Carrara, Italy: it was installed on the CMRE hybrid glider e-Folaga, and deployed in a static configuration at 1 m from the seabed in a shallow water area with 20 m of water depth. A rigid-hull inflatable boat (workboat) with hydro-jet propulsion, equipped with a GPS antenna to measure its tracks, crossed the area at an average speed of 3 m/s.

The azimuth estimate provided by the detector applied to the MVDR beamformed image along the boat crossing is shown in Fig. 6, compared to the GPS ground-truth. When approaching, the boat was detected at a maximum distance of 980 m. When leaving the maximum detection range was 600 m. This discrepancy comes from the orientation of the array in the two situations: when approaching the boat was approximately in front of the array, while, after the CPA (occurred just on top of the vehicle), it started to move behind the vehicle, so that its radiated noise was partly masked by the platform itself. Figure 7 shows the improvement achieved on elevation estimate by applying the TDOA approach, compared to the beamformer result. Both curves are compared to the GPS ground-truth. The maximum range is more limited than in Fig. 6 due to the geometry (a long and narrow acoustic waveguide prevents to resolve elevation angles at long ranges).

As we assume that the target be a surface vessel, and if we know the array depth, then we can compute the target horizontal range from elevation through a simple trigonometric formula; hence we can track the vessel with respect to the glider’s position. If the geographic position of the array is available (as it is in this case), the geographic track of the boat can be precisely estimated (see Fig. 8). Table 1 shows the maximum errors measured under these geometrical and environmental conditions on the main parameters.
Fig. 6: Azimuth estimate from MVDR beamformer, as compared to GPS ground-truth.

Fig. 7: Elevation estimates (TDOA vs. MVDR approaches), as compared to ground-truth.

Fig. 8: Boat track estimate and ground-truth in geographic coordinates.
<table>
<thead>
<tr>
<th>Parameter</th>
<th>Max error</th>
<th>Over a range of</th>
</tr>
</thead>
<tbody>
<tr>
<td>Azimuth</td>
<td>30°</td>
<td>980 m</td>
</tr>
<tr>
<td>Elevation</td>
<td>1°</td>
<td>260 m</td>
</tr>
<tr>
<td>Horizontal Range</td>
<td>40 m (16%)</td>
<td>260 m</td>
</tr>
</tbody>
</table>

*Table 1: Measured maximum errors on estimated quantities.*

5. DISCUSSION AND FUTURE DEVELOPMENTS

As a basis for preliminary proof of concept, a compact 8-element volumetric array has been used to show the improvement in positioning accuracy that can be achieved by merging a TDOA approach with the MVDR beamformer. A robust detector based on normalization has been also proposed to reduce false alarm rate.

In order to improve the achievable angular resolution, hence allowing robust detection of multiple vessels, a new array has been designed and will be used during the final PERSEUS demonstration in September 2014 [1]. It is designed for looking in all directions and it consists of a small tetrahedron of the same size as the GLASS array’s one, included in a sort of rotated pyramid with a maximum aperture of about 0.5 m. The array will be mounted on a SLOCUM glider; it will be used along with a hydrophone placed on the rear of the glider itself, in order to create a longer baseline and hence improve the capabilities of the TDOA method. This will be possible only if the glider body, which is a partly empty, long aluminium cylindrical shell, is not seen to interfere with the vessel-radiated sound in such a way to make too low the correlation between the signal sensed by the front array and that one sensed by the rear hydrophone.

6. ACKNOWLEDGEMENTS

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REFERENCES

AUTOMATED DETECTION OF FISHING VESSELS USING SMART HYDROPHONES ON AN UNDERWATER COASTAL GLIDER

Mark Wood Ocean Sonics, Great Village, NS, Canada and Ray Mahr, Exocetus Development LLC, Anchorage, AK

Abstract – AUVs operating in coastal waters where fishing vessels are also operating are at risk of becoming entangled in nets, long lines, or other fishing equipment in the water close to the surface. Autonomous gliders operating in these coastal waters are probably more susceptible to the dangers of these fishing vessels than powered AUVs, but the risks exist in both cases.

A method is proposed to minimize this risk of potential interaction with fishing gear by using two or more smart hydrophones on the coastal glider which will provide the glider with the ability to perform maneuvers that prevent interaction with gear deployed from fishing vessels. In order to command a proper glider maneuver, a fishing vessel must be detected, and classified as to type.

This paper describes both aspects of this fishing vessel avoidance issue – developing the necessary detection techniques for identifying these vessels, and the necessary performance of maneuvers that the glider must take to avoid any interaction with the fishing vessel or its gear.

Ocean Sonics of Great Village, Nova Scotia, Canada has developed a Smart Hydrophone with the ability to detect the radiated-noise signatures of different types of fishing vessels, and has the memory to store known signal characteristics of these vessels. With these stored acoustic signatures and characteristics, estimates can be made of the type of fishing vessel detected as well as the approximate range of the vessel and fishing equipment deployed. Exocetus Development LLC of Anchorage, AK has developed a glider named the Exocetus Coastal Glider which can perform a complete set of maneuvers. With this fishing vessel information, the CG control system can decide which maneuvers would be best for avoiding interaction with the fishing vessel or its gear.

Fishing vessels around the world have unique characteristics, and the proposed detection algorithms used to automatically detect different types of fishing vessels will be custom-tuned for the areas where the coastal gliders will be deployed. These algorithms will be discussed as well as ongoing work to improve these algorithms using data provided by operators of the Exocetus Coastal Glider.

Key terms – AUV, Coastal Gliders, Smart Hydrophones, Vessel Detection, Event Monitoring, Ocean Sonics, Exocetus

I. INTRODUCTION

Most underwater gliders have been designed to operate in deep ocean waters, i.e. down to 1000m, resulting in operation in waters less than 20m every 6 hours, or about 5% of the time. However, the newly developed Exocetus Coastal Glider [CG] is designed to operate in waters less than 200m, and its operation in waters less than 20m can be anywhere from 30% to 100% of the mission time, depending on the operational scenario. This later case is when the glider mission is limited to shallow waters less than 20m.

In many areas of the world, fishing vessels also operate in these shallow waters, and methods need to be developed to assist gliders in avoiding these fishing vessels and their fishing apparatus being towed. Therefore, fishing vessels create two problems for gliders: one is the
glider becoming entangled by the fishing gear, and the second is the glider colliding with the vessel, when the glider comes to the surface to communicate.

Other vessels in seas that are surrounded by land masses, e.g., Sea of Japan, Baltic Sea, Mediterranean Sea, etc. also have cargo vessels, and ferries, as well as fishing vessels, so that gliders must also be able to detect and avoid these vessels during operation in shallow waters.

This paper discusses how acoustic measurements by the glider can be used to detect and classify targets on the sea surface, as well localizing these targets. The issue related to the radiated noise of surface vessels is discussed in detail as well as the techniques used for classifying these vessels. We also discuss the characteristics of fishing vessels, cargo vessels and ferries. Furthermore, we describe the types of vessels and fishing gear that are used throughout the world.

We then propose a possible solution[s] of using Ocean Sonics’ smart hydrophones installed on the CG to avoid fishing vessels and their gear or other surface vessels using the measured radiated noise of these vessels.

II. BACKGROUND

Researchers use underwater gliders to collect in-situ data in areas of scientific interest and these measurements are related to water quality and the impact that these pollutants have on fishing stock. Scientists and fishermen have had meetings to discuss the issues of operating in the same areas and how these measurement systems [including gliders] and fishing vessels can co-exist in local, shallow waters [1].

In the USA, the latest statistics [four years ago] state that the fishing industry generated more than $185 million in sales and 2 million jobs. Estimates for the fishing industry statistics in the rest of the world are about 4 or 5 times larger than the USA [2, 3]. Hence, there is a very large number of fishing vessels operating worldwide in coastal waters. Fishing activity naturally follows fish abundance in coastal waters with only a few exceptions such as tuna, which are open water fish. The density of fishing vessels compared to ferries and cargo vessels is very dependent on the location of fishing areas, i.e. near large coastal population areas, where the three types of vessels coexist. However, in areas such as the coastal regions of Alaska, the density of fishing vessels greatly exceeds that of cargo vessels and ferries.

Fig. 1 shows an overview of the fishing areas of the world as well as the classification of the various categories of fishing vessels typically operating in these waters [4].

![Fig. 1: Fishing areas of the world](image-url)
Table 1 below defines the type of fishing activity highlighted in Fig. 1. The commercial species vary greatly and a wide variety of fishing methods are utilized to catch these species - from simple hook and line to large and elaborate nets towed by multiple ships to weirs and traps [5].

<table>
<thead>
<tr>
<th>Zone</th>
<th>Area</th>
<th>Fishing Methods</th>
</tr>
</thead>
<tbody>
<tr>
<td>I</td>
<td>Bearing Sea to 200m</td>
<td>4 species and 4 distinct methods - mid water and bottom trawl nets, long lines and deep pots -- Pollack, Cod, Halibut, &amp; King Crab</td>
</tr>
<tr>
<td>II</td>
<td>West Coast US</td>
<td>Trawl nets to 200m and purse seine to 30m for Hake, Herring and Salmon</td>
</tr>
<tr>
<td>III</td>
<td>Eastern &amp; Western Pacific</td>
<td>Tuna and deep water with seine nets</td>
</tr>
<tr>
<td>IV</td>
<td>Chatman Rise &amp; west coast Africa</td>
<td>Seine nets and deep trawls to 800-1000m for Orange Roughly</td>
</tr>
<tr>
<td>V</td>
<td>West Atlantic &amp; North Sea</td>
<td>Mid-water and bottom trawls up to 200m for Cod, Mackerel and Herring</td>
</tr>
<tr>
<td>VI</td>
<td>Pacific &amp; Indian Ocean</td>
<td>Pots and bottom trawls for shrimp and aquaculture</td>
</tr>
</tbody>
</table>

**Table 1: Fishing Zones of the World**

### III. RADIATED NOISE OF SURFACE VESSELS IN COASTAL WATERS

This section describes the radiated noise characteristics of ships that are operating in coastal waters and techniques for the measurement of these characteristics.

#### Characteristics of Ship Noise

The ability of an underwater glider to detect, classify and track a surface vessel is dependent on the radiated-noise signature of the vessel, the existing propagation conditions, the surrounding ambient noise levels [other ships such as ferries and cargo vessels or local weather conditions], and the receiving characteristics of the hydrophone system [6].

Machinery and ship propeller noise are the dominant noise sources on surface vessels when they are transiting from one port to another port. Both machinery tonals and propeller cavitation noise of vessels in coastal waters can be readily detected by hydrophones. Small ships in coastal waters typically radiate energy below 200Hz related to a ship's mechanical equipment [main propulsion or auxiliary equipment] and the dominant source above 200Hz is broadband propeller cavitation noise [7]. With the use of DEMON [Detection of Envelope Modulation on Noise] processing, not only the propeller cavitation can be easily detected, but the propeller shaft rate and blade rate can more easily be seen on the DEMON spectrum rather than on the conventional full spectrum. This leads to a very useful tool for classifying multi-targets as measured in the full spectrum [8]. Fig. 2 shows both the DEMON spectrum and the conventional spectrum for a ferry in New York Harbor with a 25-second average of the radiated-noise signature.

Diesel propulsion engines are typically found on most fishing vessels, ferries and cargo vessels, and these engines radiate energy into the water at low frequencies. These tonals are typically less than 500 Hz and are related to the rotational speed and the number of pistons on the diesel engine. Also seen in the radiated-noise spectrum are the propeller shaft rate and the propeller blade rate which is the shaft rate times the number of blades on the propeller.

#### Techniques for Collecting Radiated Noise Signatures

There are many ASW organizations that have very sophisticated methods and underwater systems for measuring the radiated noise of vessels [9]. These include fixed arrays of multiple hydrophones typically mounted on the bottom of an acoustic range, along with a precise tracking method for correcting the received hydrophone levels to a source level in dBA re
1µPa²/Hz or a dB level in some proportional bandwidth, such as a 1/3rd octave band both referenced to 1m. As expected, the amount of time spent at these sophisticated acoustic ranges determines the accuracy of the radiated noise signature for each of several operating conditions such as varying speeds and different machinery lineups.

Fig. 2: Comparison of the full spectrum to the DEMON spectrum for a ferry.

The number of these acoustic ranges with bottom-mounted arrays and sophisticated ranging systems are very few worldwide because of the cost to develop and maintain these ranges. Hence, the use of surface buoys or surface vessels with deployed vertical hydrophone arrays are becoming more popular. The depth of these arrays is dictated by the local environmental conditions and the vessels undergoing radiated-noise measurements.

With the recent interest in the impact of ship noise on mammals and other species in the ocean, techniques have been developed to collect these radiated noise signatures [of so-called ships of opportunity] with methods typically using hydrophone arrays located on the sea floor in, or near shipping lanes and near coastal regions [10, 11, 12]. The estimated radiated noise level of these ship signatures are based on the propagation loss models established for the particular hydrophone array locations.

The current challenge for collecting acoustic signals on a glider from nearby ships is more difficult if one wants to estimate the range to the target[s]. Determining the classification of the vessel if it is as close [less than 1000 meters] to the glider also requires some a priori knowledge. The importance of understanding the local waters where the glider is deployed is vitally important. This knowledge not only includes the physical oceanography of the region so that propagation loss estimates can be made, but also a detailed knowledge of the various ships i.e. cargo, ferry and fishing vessels that operate in the area. And it is the fishing vessels that offer the most danger to the gliders since they typically have fishing gear deployed.

IV. A PROPOSED SOLUTION

This section describes our proposed solution for an underwater glider avoiding fishing vessels and other vessels in shallow water [<200m] operational areas. The Exocetus Coastal Glider [CG] will be outfitted with three Ocean Sonics icListen smart hydrophones for measuring acoustic signals on the CG [12, 13]. These hydrophones will be used to measure
the noise levels on the CG and the event detection algorithms will provide estimates of the type of vessels and ranges to each of these vessels. A priori information will be stored in these hydrophones systems based on the intended operating region for the deployed CG. Use of three hydrophones will also allow an estimate of bearing to each of the detected targets.

**Coastal Glider Description**

The **Exocetus Coastal Glider** [CG] was developed in the mid 2000's and funding was provided by ONR [US Office of Naval Research]. The legacy gliders [three gliders originally funded by ONR] were all previously developed to collect data in the open ocean down to 1000m during extended deployment periods. ONR needed a glider designed specifically for coastal waters which could perform in waters with currents up to 2 knots and could easily operate in waters with large density variations. These large variations in density are due to fresh river water entering the coastal regions on all continents. These requirements led to a glider design with a 5-liter buoyancy engine which is about 10 times larger than the legacy gliders, and also allows the use of a patented adaptive ballasting control system.

Like all gliders, the CG operates employing a change in buoyancy to dive and ascend, and the fixed wings provide the lift to move forward in the water during these periods of motion. Additionally, the battery pack has been designed to provide the ability to steer the glider by rotating the hull about the longitudinal axis of the glider -- the CG of the battery pack is lower than the CG of the other components of the glider!

There are various operational modes the user may select such as station keeping within a given waypoint and radius or hovering at a certain depth and time. The **communication mode** is specified by the user and is used to determine how often he requires the CG to transmit the sensor data. Other modes available are the **surface maneuver**, which brings the CG to the surface for recovery, and the **sleep maneuver**, which turns off all power to the buoyancy system for a period of time [typically 5 minutes] to allow installed hydrophone sensors to collect data in a low noise environment. When operating in the normal glider mode, the buoyancy engine is used for the adaptive ballasting system and also for performing any trim [pitch and roll] corrections. However, these corrections use hydraulic actuators which increases the self-noise of the CG as seen on installed hydrophones.

Three additional maneuvers are designed to allow the glider the ability to avoid fishing vessels or other vessels in close proximity to the CG. These are the **emergency rise** and **emergency dive maneuvers** which command the glider to go to a particular depth and heading. The other maneuver used in the avoidance sequence is the **surface maneuver**. These three maneuvers are dictated by the CG control system based on the detection of one or more nearby vessels.

**Description of Smart hydrophones**

Smart Hydrophones are digital calibrated hydrophones with sufficient memory and processing power to analyze acoustic data in real-time. This pre-processing of data allows the selective logging of data based on fairly complex trigger events, or allow combining processed spectral data with triggered time-series data. The firmware embedded in the Smart Hydrophone can be adapted as required for new algorithms or event types. Waveform and processed data may be logged and transmitted at the same time. Short event messages are sent when event triggers are activated or end. The hydrophones have a frequency range of 10 Hz to 200 kHz, but will be configured to operate over a narrower range, up to 6 kHz. Putting the high pass filter at 10 Hz avoids swell-induced noise, but includes the very low frequency shipping sounds.

**Localization Techniques**
The Hub, used for beamforming, receives data and event messages each hydrophone in real time. The data arrives time-aligned and is stored as a multi-channel digital acoustic record.

These smart hydrophones are located on the end of each CG wing and on top of the aft vertical fin. The wing smart hydrophones are about 110cm apart and the aft smart hydrophone is midway between the two wing smart hydrophones, but 26cm aft of these units and also about 55cm above them. The design frequency of this 3-element array is about 800Hz.

With this short baseline relative to the frequencies detected, i.e. the radiated noise tonals, the localization technique uses the magnitude and phase of the event in the frequency domain. The time series is first qualified using just the magnitude, and when a valid event is detected, the phase of each hydrophone is compared to resolve the vector to the sound source for that event. Detection of multiple events is possible.

Using a pair of hydrophones for localization gives a vector to the sound source from the baseline that joins the two hydrophones. Unfortunately, there is a mirror-image vector on the opposite side of the baseline that introduces some ambiguity in the source location. This is compounded if the sound source is not on the surface. In that case, an arc of ambiguous vectors results as seen in Fig.4, where the arc ranges from the surface to the ocean bottom.

Introducing a third hydrophone, carefully located, eliminates the mirror-image ambiguity. Since the CG is only interested in targets on the surface, mounting the hydrophone on the aft vertical fin is ideal for solving this ambiguity. With three hydrophones 4 to 6 separate sound sources may be tracked.

V. FISHING VESSEL CHARACTERISTICS
The importance of knowing the types of fishing vessels in the planned operational area for the CG is vitally important! The researcher also must have a good working knowledge of all the other types of vessels, i.e. ferries and coastal shipping that the CG possibly could encounter. With this a priori information, the CG hydrophone hub processing system can be programmed to search for these potential threats.

The three potential threat to CGs from nets of fishing vessels are: 1] trawlers, 2] seiners and 3] drift nets. Of course, the fishing vessel itself is a threat to the CG. Some of the fishing methods offer low probabilities of interference due to the small cross-section of the deployed equipment. These include vessels deploying sets of hooks on long lines and pots deployed on the bottom.

Mid-water trawlers and seiners deploying their nets in water depths down to the bottom in coastal areas potentially can capture the CG in their nets. Algorithm detections must be able to classify this threat so that the CG can avoid capture by either diving or coming to the surface. When more than one fishing vessels is working together the classification is much easier than when there is only a single vessel using nets. Speed of these vessels is an important clue since trawlers usually operate at speeds of 3-5 knots continuously. The seiners typically set their
nets at about 10 knots and usually use two vessels, but once set, they retrieve their nets while stationary in the water.

Drift nets are usually deployed by one vessel at 5 to 10 knots and left alone for a while and then retrieved at about the same speed. The threat to a CG is entanglement, and the best avoidance measure is to turn away from the detected vessel.

VI. DETECTION THREATS TO A GLIDER

As mentioned previously, a complete knowledge of the local coastal waters is a must for the researcher using CGs. Both ferries and cargo vessels have well-known tracks that are followed, and the times when these vessels are operating in the area of interest is also well known.

Sources of Coastal Background Noise

The CG's hydrophones will be measuring both the self noise of the CG as well as other noise sources. The CG self noise can be a limiting factor, but one of the operational conditions is a sleep mode in which the only operational equipment in use is that of the sensors. Since the CG has an adaptive ballast control system, if water densities are varying in the operational area, the buoyancy engine can be operating, and this can be quite noisy; hence, the use of a sleep maneuver.

Ambient noise levels in shallow water are typically dominated by near shore sounds, such as harbor activity, swell crashing, crustaceans, and man-made sounds. Further from shore, low frequency sounds from distant shipping may travel for some distance if water is relatively still. Levels below 100Hz are typically from long distance shipping, whereas above that frequency wind-generated waves control the spectrum [14]. And, during times of inclement weather such as rain or sleet or snow, these levels easily dominate the wave generated noise [15]

Detection, Classification and Localization Methods

The radiated-noise signatures of various coastal vessels are obtained from various sources and estimated average levels for each type or class of coastal vessel that are expected to be in the CG operating area are inputted into the smart hydrophone hub processing system for tonal event detection algorithms.

Tonal frequencies of cargo vessels and fishing vessels are quite easy to detect as shown in Figure 5 and 6 below.

With estimates of propagation losses for the local waters, for a range of 1,000 meters or less, the received level for a vessel is established. When this level exceeds 140dB, the CG is alerted to make some maneuver to avoid this potential threat! However, propagation loss in shallow water can vary between 20 log r and 10 log r and one should use a value of about 15 log r when trying to estimate the range to a particular vessel. With the CG, the depth of the thermocline can be determined, but even with this knowledge it has been shown that tidal changes can also affect acoustic propagation in shallow coastal waters [16].
As shown previously, a second detection and classification method is with the use of DEMON processing for cavitation noise. See Figure 2. This processing element will also be located in the **smart hydrophone hub processing system**. And, it is also expected that the blade rate frequency components detected with DEMON will be higher for fishing vessels, but this hypothesis will be checked during future testing of the proposed **smart hydrophone hub processing system**.

Localization of a threat, whether a fishing vessel, a cargo ship or a ferry, will also require a bearing to that particular threat. Estimates of range can be based on received SPL of a threat, and if this range is less than 1000m then some evasive maneuver must be accomplished. Knowing the range and the bearing, the direction of the evasive maneuver by the CG can be determined. As time progresses, a bearing rate can be established for each target, and when the bearing rate in not changing or changing very little, it is time to make the evasive maneuver.

A key element of the localization method is the ability to obtain both a classification of a fishing vessel and the range, since the logic of the system must determine if the fishing vessel is transiting or is deploying nets. As stated previously, knowledge of the characteristics of the local fishing vessels is a must for the CG user!

CMRE in La Spezia has shown that localization estimates can be made with a single omnidirectional hydrophone on a glider [17]. However, use of multiple hydrophones with a beamformer will provide more accurate bearing estimates than that shown in the CMRE paper.

**Avoidance Maneuvers**
For CG survivability, three specific actions have been identified for a detected vessel: 1] No Action -- continue on defined CG mission since the threat is a long distance away, 2] Evade -- since the threat is a medium <1000m, distance away, 3] Surface or Dive -- since a fishing vessel has been detected with their nets deployed.

![CG Avoidance Logic Diagram](image)

**Fig. 6 CG Avoidance Logic**

VII. FUTURE EFFORTS

Future testing of the CG and the smart hydrophone hub processing system is recommended and will be accomplished by the team of Exocetus and Ocean Sonics. This testing will be accomplished in both Alaskan and Nova Scotian waters since these two areas are the locales of the two companies.

Simulations using data previously collected in the field will allow controlled testing of improved avoidance algorithms.

Self noise testing of the hydrophone system on the CG is required to determine the time needed for the sleep maneuver. The ability of the processing system to determine classified threats and track these threats in the presence of one or more targets will be verified using the simulation data, and ultimately live field data.

VIII. CONCLUSIONS

A proposed method for the automatic detection of fishing vessels has been discussed and the maneuvers needed for the CG to avoid interference has also been discussed. The proposed approach seems achievable, but needs to be tested thoroughly.

**References**


NEXOS OBJECTIVES IN MULTI-PLATFORM UNDERWATER PASSIVE ACOUSTICS

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\textbf{Abstract:} The objective of the NeXOS project is to develop cost-effective, innovative, and compact multifunctional sensor systems in ocean optics, ocean passive acoustics and for an Ecosystem Approach to Fisheries (EAF), which can be deployed from mobile and fixed platforms, with data services contributing to the GEOSS, the Marine Strategy Framework Directive (MSFD) and the Common Fisheries Policy of the European Union. The development of innovative hydrophones will focus on the pre and post-processing of acoustic information and improved transducer integration, reducing size and overall procurement and operations cost while increasing functionality. An important part of the effort will focus on the need for greater dynamic range and the integration on autonomous platforms, such as gliders and profilers. Embedded processing will be reconfigurable, allowing for the monitoring of MSFD Good Environmental Status descriptors I (Biodiversity) and II (Underwater Noise) as minimal requirements. The first phase of the project consists in interacting with scientific communities and the industry in order to narrow down initial requirements and possibly extend the planned functionalities to new applications. The presentation will provide an overview of the project and an update on current progress, with a focus on unmanned vehicles and mobile platforms more generally.

NeXOS is co-funded by the European Commission 7\textsuperscript{th} Framework Programme, the Ocean of Tomorrow 2013.
1. INTRODUCTION

The objective of NeXOS is to develop cost-effective, innovative and compact integrated multifunctional sensors for ocean optics, ocean passive acoustics, and sensors for an Ecosystem Approach to Fisheries (EAF), which can be deployed from mobile and fixed ocean observing platforms. The project has several transversal objectives for these sensor systems. An example is to develop sensors downstream services for the GEOSS and the Global Ocean Observing System - GOOS, in order to contribute to the monitoring of European marine waters’ Good Environmental Status (GES) (as laid down in the Marine Framework Strategy Directive). This paper introduces the general framework for NeXOS and the background rationale, then focuses on the passive acoustic development objectives.

2. EU POLICY AND OCEAN OBSERVATION

The European Union, with contributions from Member States and the European Commission, has developed the Marine Strategy Framework Directive\(^1\) (MSFD) to support a sustainable and integrated approach to the monitoring of the ocean. This Directive includes initiatives to satisfy the Good Environmental Status (GES) of marine waters (also linked to the Water Framework Directive), the Data Collection Framework for Fisheries\(^2\) (DCF), which aims to improve management by collection of high-quality data, the INSPIRE Directive\(^3\), which aims to improve the accessibility of standardised datasets, and the Blue Growth communication, which addresses growth opportunities for sustainable marine sectors\(^4\). These policies have been supported by various projects, initiatives and infrastructures at European level (ESONET, EMSO, EuroArgo, Eurosites, JERICO, Seadatanet, GROOM, HYPOX, COOPEUS, Ferrybox, etc.), with contributions from several ocean observing communities, including partners involved in the NeXOS project. NeXOS intends to turn the resulting functionalities identified into technical requirements for the sensors to be developed.

3. THE RATIONALE FOR NEXOS SENSING TECHNOLOGIES

Ocean in-situ sensors can generally be classified into three general transduction mechanisms. Either they use optics, acoustics or electro-chemical techniques [1]. Optical

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4. COM(2012)494
and acoustics transducing methods are increasingly attractive and identified as promising candidates for innovation, variable coverage and adoption by future end-users. These methods notably have potential to reduce costs due to their capacity to be multifunctional (single sensor systems addressing several parameters), be deployed on a large majority of ocean monitoring systems, from surface to the seafloor, and be operated for long periods with potential for less maintenance needs. They also have good potential to contribute to GES descriptors (Error! Reference source not found.). These two techniques are complementary and cost-effective methods for in-situ continuous ocean measurements and sensing of ocean key variables [2]. Optical and acoustical probes can also be added to small and low-cost platforms [3].

With respect to fisheries management and NeXOS specific contribution to the CFP, new, very sturdy, small size and very-low cost sensors will also be developed specifically for fishing vessels. New chlorophyll and oxygen sensors (optical) will be developed for installation on fishing nets. The sensors, defined herein as an EAF sensor system (Ecosystem Approach to Fisheries), will be integrated on the RECOPESCA system, promoted and led by Ifremer. Specific communication needs and sturdy packages will be required for the new sensors to be added to RECOPESCA, and thus NeXOS dedicates specific activities for their respective developments. Other NeXOS developments will be studied for potential integration within the system. The sensors will also enhance the contribution of fishing vessels to the monitoring of several GES descriptors.

### Table 1 Coverage of NeXOS Sensor Systems for the Marine Strategy Framework Directive (MSFD) Good Environmental Status (GES) Descriptors

<table>
<thead>
<tr>
<th>NeXOS Development</th>
<th>Optical Sensors Systems</th>
<th>Passive Acoustics Sensor Systems</th>
<th>EAF Sensor System</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>MSFD GES Descriptors (related variables)</strong></td>
<td>Descriptor 4: Elements of marine food webs (e.g. Phytoplankton, Nutrients, etc.)</td>
<td>Descriptor 1: Biological diversity (e.g. Aquatic mammals, fish reproduction areas)</td>
<td>Descriptor 1: Biological diversity (e.g. Identification of Fish catch, depredation)</td>
</tr>
<tr>
<td></td>
<td>Descriptor 5: Eutrophication (e.g. O2)</td>
<td>Descriptor 11: Introduction of energy, including underwater noise (e.g. Ambient and anthropogenic noise sources)</td>
<td>Descriptor 3: Population of commercial fish / shell fish (e.g. Identification of Fish catch, geographic distribution)</td>
</tr>
<tr>
<td></td>
<td>Descriptor 8: Contaminants (e.g. PAHs, Turbidity, Acidity)</td>
<td></td>
<td>Descriptor 5: Eutrophication (e.g. O2, Chlorophyll)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>Descriptor 7: Alteration of hydrographical conditions (e.g. Changes in temperature and salinity)</td>
</tr>
</tbody>
</table>

**4. NeXOS Sensors Synergies**

The developments of NeXOS are based on an integrated strategy for ocean monitoring. Several sensor systems will be developed, tested and validated for specific technologies and monitoring strategies (i.e. ocean optics, ocean passive acoustics, and EAF monitoring), and will provide an integrated, technologically coherent system for multi-scale, multi-parameter monitoring of the oceans. This will inevitably provide potential for...
synergies. Some examples are provided below, these and others will be further studied in an early stage of the project to optimise requirements:

- Optical sensor systems can measure chemical compounds and biologically relevant parameters, such as nutrients, dissolved gases, acidity and organic matter. Nutrients and dissolved oxygen are particularly relevant to fisheries management as they have a fundamental role in the production of phytoplankton, at the base of the trophic chain.
- Passive acoustics used for noise and marine mammal acoustics, also have potential in the identification of fish reproduction areas (descriptor 3 of the MSFD) and the non-lethal mitigation of catch depredation by whales in fisheries activities.
- The Ecosystem Approach to Fisheries (EAF) sensor system will add chlorophyll and oxygen to the currently collected data on fish, fishing vessel activity, salinity, temperature and depth. The improvements made by NeXOS in the compactness and multifunctionality of optical sensors and passive acoustics sensors should be assessed for integration on the EAF sensor system, for the monitoring of depredation from marine mammals.
- While sound travels over large distances in the ocean, light is absorbed at short distance and cannot be used for passive underwater remote sensing in practice. In contrast to ocean optics, passive acoustics can be used to measure a range of variables and can be used for remote sensing at geographic scales from meters to hundreds of kilometres. Acoustic techniques are therefore complementary to optics in multidisciplinary ocean observing systems for scanning the ocean at a distance.

5. Sensor Web

The ‘Marine Knowledge 2020’ communication of September 2010 showed that improved management of marine observations and data would reduce the cost of operations at sea, stimulate innovation and reduce uncertainty in knowledge of future oceanic conditions (IP/12/920). At present data are generally held at institutions throughout Europe, making it difficult e.g. to find data for a particular parameter in a specific area. Obtaining access and authorisation is problematic, and it is time-consuming to collate heterogeneous datasets from different sources to form a coherent picture. Many potential activities cannot obtain funding due to the high costs that must be met by marine operators. The European GMES/MyOcean programme has now set up a marine service using satellite and in-situ data to provide oceanographic forecasts, and the EU's Data
Collection Framework has established a process for a structured collection of fisheries data. Many high value services, such as management of fish stocks and the protection of coastal infrastructures, will be provided only by a few organizations unless the accessibility of marine data improves. This situation is non-competitive and inefficient. The same applies to passive acoustics, thus some initiatives are also taking place in order to canalise data through different portals, like the ESONET/LIDO demonstration mission and the Quiet Ocean experiment. However, these initiatives also need interoperable solutions for exchanging data of scientific value. With improved interoperability, small businesses and young engineers would also be able to develop new products and services using a variety of data from different sources. NeXOS will provide the tools to facilitate the task of data managers and managers of observation systems to introduce interoperability principles. For all new sensors, NeXOS will adopt OGC's Sensor Web Enablement standards to enable Web-based sharing, discovery, exchange and processing of sensor observations, and operation of sensor systems. Data distribution and access will be easier both for the provider and end user. This will support innovation and the development of new services and applications.

6. PASSIVE ACOUSTICS IN NEXOS

The MSFD and the increasing need to monitor underwater noise for Environmental Impact Assessments of marine activities have increased demand for cost-effective multipurpose instrumentation. Although regulations are now in place for underwater noise (MSFD Descriptor 11), marine acoustic sensors and acoustic data processing remain costly. This is mainly due to procurement and operational costs, including the need for experts at nearly each stage, from sensor operation through to data processing. Recently a European task group on underwater noise was formed and proposed the establishment of a set of underwater noise observatories as a monitoring solution to GES Descriptor 11/Underwater Noise, as well as noise indicators [6]. In addition to monitoring noise, new sensors and platforms can provide cost-efficient information for the assessment of marine mammal populations (MSFD/GES descriptor 1), as was demonstrated with T-POD porpoise detectors [7] and highlighted in [8], C-POD, among others. There are many further potential applications of these sensors, including the detection of fish reproduction areas (relevant to descriptor 3 of the MSFD, and the CFP) [9-10], the non-lethal mitigation of catch depredation by whales in fisheries activities [11], detection of Green-House Gases (GHG) seeps from pipelines and deep sea carbon storage, gasification of methane clathrates [12], estimation of rainfall, detection of low-frequency seismic events, ice-

![Fig. 2 Ocean soundscape diagram with typical natural and anthropogenic sources. The two heavy-lined curves show (in black) a typical ocean ambient noise power spectral density and (in blue) The light blue boxes (whose vertical size and position is arbitrary) show the frequency range of natural sources, from [4] and [5].](image-url)
cracking, ocean basin thermometry and tomography [13], acoustic communication. Fig. 2 provides a spectral representation of the large variety of sound sources in the ocean.

Current oceanic passive acoustic sensor systems generally include one or several transducers (hydrophones), signal conditioning, instrument interface with communication and control capability, and internal or external power supply. In stand-alone platforms, passive acoustic data are generally stored for later recovery and analysis, unless (rarely) there is a direct RF link to shore. Data may also be sent directly through cabling to a host, whether it is a ship or vessel of opportunity (towed array, [14]), or a cabled observatory shore station [15-16]. Most ocean monitoring platforms are still stand-alone with an RF link of limited bandwidth, and the growing use of glider and profiler technologies in global observation will continue to increase the use of RF transmission, mainly via costly and energy-demanding satellite links. Most commercial passive acoustic sensors on the market are also unable to perform simultaneous measurement of sound level extremes (very low and very high), e.g. most sensors are not able to respond to the European Seas Observatory Network (ESONET) label requirements of 50 to 180dB re. 1μPa, and data processing has to be performed on costly and/or bulky systems, generally impractical for mobile platforms.

Within NeXOS the following technical progress will be achieved:

- The new sensors analog front-end will be designed to reduce input noise and will result in the ability to measure below Sea State Zero and very high sound levels simultaneously.
- Embedded pre-processing of data, customised to small low-power platforms. Measurements will include several sources, such as ambient noise, anthropogenic noise, bioacoustics sources, and where applicable, ancillary variables such as salinity, temperature and depth. Data pre-processing will build on available open-source software.
- Use of reliable low-cost, low-power components and technologies for deployment on open-ocean low-power platforms and compactness for integration on mobile platforms. This will be performed through smaller integrated circuits for signal conditioning and interfacing.
- Data and sensors will be web-enabled, with sufficient metadata to warrant traceability.

Sensors will be calibrated according to the most recent standards [17]. This will result in securing broader market penetration, in particular in the field of noise monitoring (GES Descriptor 11). The requirements and engineering will be further assessed and specified in the first year of the project. A new compact sensor system will be integrated and validated on several fixed and mobile platforms, including the NKE PROVOR profilers (or
Profiling floats) and the ACSA-ALCEN SeaExplorer glider. The operation of these platforms equipped with these multifunctional passive acoustic sensors will provide novel information on the evolution and management of the ocean soundscape, as well as efficient low-cost tools for measuring acoustic impacts in marine EIAs.

7. ACKNOWLEDGEMENTS

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THE FUSION OF DIGITAL TERRAIN MODELS MEASURED FROM MULTIPLE ACOUSTIC SENSORS – APPLICATION TO THE DAURADE AUTONOMOUS UNDERWATER VEHICLE

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Abstract: Building an accurate digital terrain model (DTM) of the seabed is a key issue for various military and civilian hydrographers applications. In the past decades, the emergence of autonomous underwater vehicles (AUV) offers new methodologies to collect the bathymetric data used in the estimation of the DTM. In our study, we use the DAURADE AUV platform which is capable of acquiring bathymetry with two acoustic sensors: A multibeam echo sounder (MBES) and an interferometric sidescan sonar (ISSS). The two sensors (MBES and ISSS) are synchronized to operate concurrently. In fact, the final DTM can be improved by performing a fusion of the data; the two systems acquire the bathymetry with different resolutions, geometries and error models; these parameters are introduced in the fusion process to improve the estimation of the DTM and to increase its accuracy.

The aim of this paper is to describe the fusion method and discuss our preliminary results on modeled data. First, the modeling of two acoustic sensors (MBES and ISSS) will be briefly described. The input data sets are simulated by applying the sensor models on simplified seabed models. The use of seabed models provides ground truth and, therefore, allows for quantifying the accuracy of the fusion process.

Keywords: AUV, multibeam, interferometric sidescan, fusion, bathymetry, seafloor.
1. Introduction

One of the major issues of the international hydrographic community is the question of the irreducible uncertainty in modern surveys to build an accurate digital terrain model (DTM). As the base data for DTM estimation is a huge amount of soundings with measurement noise, automatic data-cleaning and DTM production package have been recently developed such as the Combined Uncertainty and Bathymetry Estimation (CUBE) [1] or the Cleaning through Hierarchic Adaptive and Robust Modelling (CHARM) [2]. These two algorithms process the full coverage multibeam data with every sounding including an estimate of its uncertainty (CUBE case). Although the quality factor proposed by [3] can be used to model uncertainty, these algorithms can’t integrate heterogeneous and qualitative data like expert opinion or bathymetry derived from shape from shading method. When many bathymetric data with different spatial resolution, coverage and uncertainty are available for the same area, the question arises whether this redundancy and complementary can be fused to generate a better DTM.

In the past decades, the emergence of autonomous underwater vehicles AUVs which can be equipped with a wide variety of acoustic sensors or sonar systems, offers different methodologies to collect bathymetric data. In shallow water and for full coverage area survey, the two most used systems are multibeam echo sounder (MBES) and interferometric sidescan sonar (ISSS). The MBES still the standard sonar used for accurate hydrographic survey. But when installed on AUVs, which navigate usually close to the seafloor, it will suffer from its limited angular coverage and thus a lot of time consuming for full coverage which is difficult for vehicles with limited battery autonomy. In such survey conditions, there are advantages to use ISSS systems. An ISSS has a swath width of more than 10-times the altitude of the sonar and produce high resolution bathymetry across track. This would reduce significantly the time of the survey for full coverage. On the other hand, such system suffers from many disadvantages. The geometry of ISSS transducers doesn’t allow gathering data in nadir area. It has a limited bathymetric accuracy about 2-3% of water depth. Another issues are baseline decorrelation and the shifting footprint effect as described by Lurton [4]. In spite of these significant disadvantages, recent advantages in system electronics and algorithms have improved ISSS performance. In many AUV survey missions such as detecting and mapping submerged wrecks, rocks, and obstructions, the fuse of bathymetry derived from MBES and ISSS would improve productivity.

In this paper, we focus on MBES-ISSS bathymetric data fusion under uncertainty measurement. The remainder of this paper is organized as follows. Section 2 describes the bathymetric fusion model. Section 3 introduces the modeling of two acoustic sensors (MBES and ISSS) and the performance of the fusion model.

2. MBES-ISSS bathymetric data fusion model

Digital surfaces, derived from different sensors, contain intrinsic error due to acquisition and processing methodology in relation with terrain type and shape. In order to overcome limitations of each DTM, an intelligent fusion which considers uncertainty and reliability of each sensor is required. In radar community, the most used fusion algorithm to combine DTM’s (SAR interferometry, LIDAR...) is a weighted average of inputs in each grid cell. As weights factor are not usually available, data accuracies are estimated from DTM (roughness, slope,...). To be robust to blunders, other methods are used by representing local patches as a sparse combination of basis patches [8].
These algorithms can’t integrate prior knowledge about the precision and reliability of sensors which can vary with time and environment conditions. To deal with such kind of measurement, many theories have demonstrated the capability of modeling the uncertainty. We can mention imprecise probability, possibility theory and theory of belief function.

The theory of belief function, also known as Dempster-Shafer Theory (DST), was developed by Shafer [6] and initiated by the work of Dempster on imprecise probabilities. It's one of the popular approaches to handling uncertainty in the data fusion literature and it's often considered as a generalized model of probability and possibility theories. We will not introduce the basic of this theory. The interested reader can find sufficient interpretations of evidence theory in the literature [10].

In our case, inputs are soundings \( z_i \) with known position \((x_i, y_i)\) and standard deviation \(\sigma_i\) processed from MBS and ISSS and we look for more accurate \( z_i \) values by combining them. In [7] Petit-Renaud and Denoeux propose an evidential regression (EVREG) analysis of imprecise and uncertain data. In this model, evidential theory are extended to fuzzy sets where focal elements are fuzzy variables. The basic idea is to construct fuzzy belief assignment (FBA) in two steps: discounting FBA’s \( m_i \) according to a measure of dissimilarity between inputs vectors, and combination of the discounted FBA’s [7]. The model in our case may be summarized as follows.

Given a set of \( n \) sounding values \((y_i, z_i, \sigma_i, p_i)\), at a ping along track position \( x_i \), a FBA \( m_i \) can be defined for each pair \((y_i, m_i)\) as:

\[
m_i(F_i) = p_i
\]

\[
m_i(Z) = 1 - p_i
\]

Where \( F_i \) is a Gaussian fuzzy number with center \( z_i \) and standard deviation \(\sigma_i\) and reliability \( p_i \) of the sonar. Each element \( e_i \) of the inputs \( I = \{e_i \mid e_i = (y_i, m_i), i = 1,2,...,n\} \) is a piece of evidence concerning the possible value of \( z_i \), which can be represented by a FBA \( m_z [y, e_i] \) as a discounting of \( m_i \):

\[
m_z [y, e_i] = \begin{cases} 
m_i(A)\varphi(||y - y_i||) & \text{if } A \in F(m_i)\backslash\{Z\} \\
1 - \varphi(||y - y_i||) & \text{if } A = Z \\
0 & \text{otherwise}
\end{cases}
\]

Where \( \varphi(.) \) is a decreasing function from \( \mathbb{R}^+ \) to \([0,1]\) verifying \( \varphi(0) \in ]0,1[ \) and \( \lim_{d \to \infty} \varphi(d) = 0. \) \( \varphi(.) \) represent a discounting function that measure the dissimilarity of the variable of interest \( z \) using a suitable metric \( ||.|| \) between input vectors \( y \) and \( y_i \). If \( y \) is close to \( y_i \), \( m_z [y, e_i] \) and \( m_i \) are very similar and vice versa. When the metric \( ||.|| \) is defined as Euclidian distance, a natural choice for \( \varphi(.) \) is [7]:

\[
\varphi(d) = \gamma \exp(-d^2)
\]

Where \( \gamma \in ]0,1[ \) is a real parameter (usually taken \( \gamma \in ]0.95,1[ \) ).

The information provided by each element of the input set can be combined by the conjunctive rule of Dempster. In practice we can neglect the effect of inputs \( y_i \) far from the position of interest \( y \) and only take \( k \) nearest neighbours. The final FBA is then:
The presented EVREG model is applied for each sensor, and their outputs are combined using Dempster’s rule to form a new FBA \( m_i = m_i^{+1} \oplus m_i^{-2} \). The probabilistic density \( BetP[y, I] \) associated to \( m_z[y, I] \) exits, and has the following expression:

\[
BetP[y, I](z) = \sum_{A \in \mathcal{F}(m_z[y, I])} m_z[y, I](A) \frac{A(y)}{|A|}
\]

(5)

Where \( m_z^+[y, I] \) is the normalized version of \( m_z[y, I] \) and \(|A|\) is the cardinality of A.

For point prediction of the z value we can use the center of gravity of A \( (z_A^*) \). Then \( \hat{z} \) can be expressed as:

\[
\hat{z}(y) = \sum_{A \in \mathcal{F}(m_z^+[y, I])} m_z^+[y, I](A) z_A^*
\]

(6)

To measure the uncertainty involved in the prediction of FBA, we can used the measure of nonspecificity generalized for belief functions in [9]. This is defined as:

\[
N(m_z[y, I]) = \sum_{A \in \mathcal{F}(m_z[y, I])} m_z[y, I](A) \log_2 |A|
\]

(7)

3. Experiments

3.1 Simulated data

To validate our fusion process, we have developed a simulator for MBES and ISSS sonar. The simulator is based on construction of an adequate scattering-point model using a facet approach. In this approach, the synthetic bottom DTM is modeled as a set of triangular facets whose center represents the location of a scattering point and each facet has a unique identity composed of its normal vector, surface and its amplitude depending in sediment type. The signal received for each facet takes the form:

\[
S_r(t) = A \cap T \ e^{(j2\pi\nu_0(t-\tau_i))}D_e(\varphi)S_IBS_0(\theta) - \frac{2\alpha r_i}{r_i^2}
\]

Where \( S_e(t) = A \cap T \ e^{(j2\pi\nu_0 t)} \) is the emitted signal, \( \tau_i \) two way time propagation, \( D_e(\varphi) \) the directivity of emission antenna, \( S_I \) triangular facet surface, \( BS_0(\theta) \) is the backscattering strength depending on incident angle and sediment type, \( \alpha \) is absorption coefficient and \( r_i \) is sonar facet distance.

A signal of 300 pings of a synthetic seafloor with three sediment types (flat mud, sand waves and rock) are collected with a MBES and ISSS with the following characteristics:

**MBES**
- 100 kHz system
- Beam aperture along-track 1°
- Beam aperture across-track 1°
- 64 transducer element

**ISSS**
- 100 kHz system
- Beam aperture along-track 1°
- Maximum range 75 m
- 3 receiving transducer with 25° depression angle

For the MBES the process consist of beamforming, depth detection by centre of gravity of the amplitude envelope and zero-phase difference instant estimation. For ISSS, the phase difference direction estimation is done by the so-called Vernier method using the three pairs of receivers. Next an outlier filter is applied to the estimated arrival angles. To estimate receive angle uncertainty, a 2\(^{nd}\) degree polynomial is fit to each horizontal range.
bins of 30cm size and only one soundings is retained per bins. Sounding uncertainty of each sounding is estimated by using the quality factor proposed by Lurton et al. [3, 5]. It is the ratio between the estimated sounding and its standard deviation obtained from signal characteristics.

3.2 results

The estimated soundings data for both systems is collected, using the simulator described above, throw three survey lines. Line in position y=0 is surveyed with both systems and the two other only with MBES (see ‘FIG.1’).

Figure ‘Fig.2’ displays the results obtained for three scenarios data fusion: (a) ISSS alone, (b) ISSS and MBES survey line at y=0 and (c) ISSS line at y=0 and three MBES lines at y=0, 60 and -60 meters. The pignistic probability distribution, also the width of first and ninth deciles interval, can be seen to reflect the uncertainty taking into account both the scatter and the density of input soundings. The uncertainty is maximal in the extremity of the profile in all scenarios because there is no input data. It’s also maximal for the two first cases in the -67 m range because of shadows in ISSS data. This effect can be also detected on the nonspecificity measurement. The nonspecificity measurement shows the contribution of the fusion of the MBES and ISSS data by the decreasing of uncertainty in overlapping ranges.

REFERENCES

Fig. 1: Simulated sea bottom and survey lines (dashed lines).

Fig. 2: One ping fusion result for three scenarios. Left figures show the estimated depth at the same resolution of the ISSS (red dashed lines) with the actual depth (black dashed lines) and first and ninth deciles of the pignistic distribution (grey area). Right correspondent nonspecificity measurement.
IMPROVEMENT OF AUV-BORNE SEABED MAPPING WITH QUALITY MAPS USING STATISTICAL ANALYSIS

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Abstract:

Sonar data is commonly affected by noise due to the processing of scatter signals and interference of acoustic waves scattered from the seabed. To overcome this problem and to limit the noise in sonar images, the sonar operator can change the sonar settings (e.g. range, pulse length, modulation, inter-track distance, etc.) to acquire the best possible acoustic data. On board autonomous underwater vehicles (AUV), due to the low bandwidth of the communication with the robot, the real time definition of the best settings by an operator is nearly unfeasible. For these reasons, we have developed an analysis method for automatically assessing the quality of the data. The results of this process are then sent to the AUV planning module which can change the sonar settings (e.g. inter-track distance).

The classical approach is based on the correction of the artefacts related to the wave propagation in water column and the characteristics of the sonar system. This approach requires strong a priori knowledge of the system and the conditions of acquisition of the sonar data.

The main objective of this paper is to propose a statistical measure of quality of the sonar data acquired using AUVs. This statistical measure would be representing a quality map for the input sonar data. As no prior measurement of similarity or dissimilarity of sonar images is given, the decision to whether accept the quality of data as noisy/non-noisy will be based upon statistical hypothesis testing. To accomplish the quality mapping, spectral domain filtering is performed to extract the residual image representing the speckle. Based on Maximum Likelihood (ML) method, parameters are estimated from the data for Rayleigh distribution and its fit is evaluated using Goodness-of-fit (Gof) test.

Experimental results show the viability of the proposed approach while mapping the data into quality matrix representing the acceptable regions on sonar data acquired using DAURADE.

Keywords: Quality measure, Cartography mapping, Noise models, Sonar data, AUV (Autonomous Underwater Vehicle)
1. INTRODUCTION

The Autonomous Underwater Vehicle (AUV) is becoming a common vehicle to collect acoustic images of the seabed. The sonar mounted on the vehicle sends ultrasonic pulses in a defined direction and records the signal resulting from the interaction on the transmitted pulse and the environment (scattering, reflection, attenuation, etc).

Coherent imaging systems (like synthetic aperture radars, sonar, ultrasound and laser imaging) are commonly affected by multiplicative noise (also known as speckle noise). The speckle noise in these systems is caused by the addition of coherent and random interference in the backscatter signals [1]. A trivial solution to suppress the noise is to directly apply the de-noising low-pass filter, but by directly applying the de-noising filters, relevant information can be lost which significantly affects the later processes of object detection, identification and recognition.

Generally, the speckle noise can be represented by: \( D(x, y) = I(x, y) \ast r(x, y) \), where \( D(x, y) \) is the observed sonar image, \( I(x, y) \) is the original image and \( r(x, y) \) is the multiplicative component of the speckle noise. The de-speckling filters can be categorized into two main groups: Statistical based and Frequency based filters [2]. The known de-speckling filters explained in [3] (like Lee, Frost and Kuan) are statistical based filters which used a-priori statistical models of speckle noise. Frequency domain filters (like wavelet and Fourier) are non-adaptive filters which take all the signal components into consideration and then process the data [4, 5].

All these de-noising filters smooth out the noise while retain the features in the image but cannot provide any qualitative assessment about the data before and after filtering. Since no prior information is given to describe or estimate the quality of sonar image, a sonar image can be describe as a good quality image if the seabed features appear very well just like they would have been observed under the ideal operating conditions of the sonar system [6].

The main objective of this work is: quantitative analyses of the sonar image by providing a quality map of the data into good and bad regions along the track of AUV in different ranges of sonar image. This transformation of quantitative sonar data into quality map would give to the AUV, a better understanding on how the seabed is acquired by the sonar and would ultimately lead to time and costs saving, by optimal real-time mission re-planning.

The rest of the paper is organised as follows. In section 2, the proposed approach for quality mapping is discussed. Experimental results based on the proposed approach over sonar data is given in section 3. Concluding remarks and future prospects are provided in section 4.

2. PROPOSED QUALITY MAPPING APPROACH

The information content in the sonar data in terms of quality degrades significantly across the range. It is widely accepted that the noise in the sonar data can be appropriately modelled by the Rayleigh distribution [7]. This can be exploited in the real case scenarios where the data is, acquired through DURADE AUV, having different scale of noise in different ranges of the swath. A hypothesis about the quality of data can be set on the basis of the Rayleigh distribution and the goal of making these inferences can be achieved by
testing the hypothesis. The proposed approach can be summarized into the following steps:

- First, the sonar image is partitioned into equal sized windows and each image window is transformed from the spatial domain into frequency domain using Fourier transform.
- Apply the Butterworth high-pass filter in the frequency domain image, which gradually transition from 0 to 1 to keep high frequencies outside a radius and discard the low frequency values inside the radius.
- Calculate the residual image by applying the inverse Fourier transform which contains only the speckle noise representing the high frequency components in the image.
- Estimate the parameters for the Rayleigh distribution from the data using the Maximum likelihood estimation (MLE) method.
- Using hypothesis testing (goodness of fit test), measure how well the Rayleigh distribution fits to the observed residual data.
- Based on the probability of the support in the goodness of fit test, the input sonar data is mapped into a quality matrix.

In the remaining section, the detail of the proposed approach is explained and justified.

A. Spectral Domain Filtering and Analysis:

The amplitude and the phase of the backscattered signal recorded on the sonar transducers are statistically independent. The amplitude is a function of the object reflectivity and the phase is a function of the surface shape. In the domain of image processing, the Discrete Fourier Transform (DFT) is widely used in numerous applications like image analysis, enhancement, filtering, reconstruction and compression [8]. In Fourier transformed image, the low frequencies correspond to the slowly varying information (e.g., homogeneous areas), while the high frequencies correspond to the quickly varying information (e.g., edges). The speckle noise components in the input sonar image mostly belong to the high frequencies in the Fourier space, thus the information content about the speckle can be obtained by filtering the low frequency components in the transformed domain.

The Butterworth filter has the property to gradually suppress the frequencies, where the roll-off (sharpness/slope of the transition from the pass-band to the stop-band) is controlled by the filter order [8]. The Butterworth high-pass filter keeps the frequencies outside a radius \( r_0 \) and discards those values inside the radius \( r_0 \). The high-pass Butterworth filter is given by:

\[
H(u, v) = \frac{1}{1 + \left( \frac{r_0}{w(u, v)} \right)^{2n}}
\]  

(1)

Where \( w(u, v) \) denotes the distance from the centre of the spectrum, \( r_0 \) denotes the cut-off frequency which controls the radial size of the filter and \( n \) denotes the order of the filter which controls the transition from stop-band to pass-band (i.e. from 0 to 1). A family of filters can be created by varying \( n \) to increase or decrease the slope \( r_0 \). The Fourier domain image \( F(u, v) \) is multiplied with the Butterworth high-pass filter \( H(u, v) \) of same size, to produce a filtered image given by:

\[
Z(u, v) = F(u, v) \ast H(u, v)
\]  

(2)
The filtered image $Z(u, v)$ contains the speckle noise components representing the high frequency components in the given sonar image.

Beside the issues related to the configuration of sonar systems, the unresolved problem in any sonar data quality mapping is the lack of substantial prior information or ground truth sonar image which can be used to make a comparison for quality assessment. In the case of limited prior information related to the noise, a quality mapping can be achieved. The only information about the noise can be obtained from residual image $R_t(x, y)$, which is computed by taking the inverse DFT of the filtered image $Z(u, v)$. The residual image only contain information about the speckle component in the sonar image as the low frequency component representing the homogeneous areas in the sonar image have gradually been removed. This operation is very important for understanding the behavior of the noise, as the noise remain the same even in different regions of the seabed’s, therefore only their distribution can be used to find their model.

B. Quality Mapping: Model Estimation and Goodness-of-Fit Test

The Rayleigh distribution is widely used to study the speckle noise in coherent imaging systems. The probability density function of the Rayleigh distribution is given by [9]:

$$s(x; \sigma) = \frac{x}{\sigma^2} e^{-\frac{x^2}{2\sigma^2}}, x \geq 0, \quad (3)$$

where $\sigma > 0$, is the scale parameter of the distribution. The cumulative distribution function is given by:

$$S(x) = 1 - e^{-\frac{x^2}{2\sigma^2}}, \text{ for } x \in [0, \infty) \quad (4)$$

In order to understand the behaviour of noise in the residual image, it is very important to model its distribution. For modelling any distribution, we need to find or estimate the parameters of the assumed distribution. The objective is to identify the good parameters for the Rayleigh distribution that is mostly likely to have generated the speckle vector $R_t$. The two most commonly used methods for parameter estimations are the least-square estimation (LSE) and maximum likelihood estimation (MLE) [10]. We choose to estimate the model parameters using MLE because it is more useful in hypotheses testing or constructing confidence intervals and inference in statistics.

Goodness-of-fit (Gof) techniques examine how well a sample of data agrees with a given distribution as its population [10]. Some important Gof tests are; (i) chi-square tests, (ii) moment ratio, (iii) correlation based tests, (iv) empirical distribution function based tests. The Gof procedure defines a test statistics, which measure the distance between the hypothesis and the observed data, and then calculate the probability of obtaining the data, assuming the hypothesis is true. The smaller probability will indicate poor fit, while high probability corresponds to the good fit. Due to the intrinsic restrictions (due to the size of data, distribution type etc.) on other Gof tests like chi-square tests, we exploit the Kolmogorov-Smirnov (KS) test which is based on the empirical cumulative distribution function. The KS test computes the largest difference between the theoretical and the empirical distribution function [11]. Assuming that the random variable $R_t$ represent the residual data and $S_t$ represent the model distribution then the two sample KS test can be
used to test whether the two underlying probability distributions differ. The KS statistics is given by:

\[ M_t = \max |R_t(x, y) - S_t(x, y)| \]  

(5)

Where \( M_t \) is the least upper bound of all point wise differences \(|R_t(x, y) - S_t(x, y)|\).

If the sample comes from the same distribution then \( M_t \) converges to 0. The hypothesis regarding the residual data is rejected if the test statistics \( M_t \) is greater than the critical value.

The sonar data correspond to good quality (features appear very well), if the noise distribution in the data follows the Rayleigh distribution. If we assume that our residual measurements are governed by a particular distribution, we can make two working hypotheses about the distribution: The null hypothesis: \( H_0 \): The observed distribution follows the Rayleigh distribution, while the alternative hypothesis: \( H_A \): The observed distribution do not follow the Rayleigh distribution. The qualitative map must provide a quantitative value with confidence in the acceptance or rejection of the noise model. If the difference is less than a determined value then the agreement is satisfactory, but if the difference is much greater than the determined value then it is not satisfactory.

Based on the Kolmogorov-Smirnov test statistics given in equation (5), the null hypothesis regarding the observed distribution is rejected if the test statistics, \( M_t \) value, is greater than the critical value (p-value) obtained from table [12]. The higher p-values represents the lower distance \( M_t \) value at the low range of the image, while the low p-values represent the higher distance \( M_t \) value at the far range of the residual image. Based on the p-value the sonar image is mapped into three different regions colored into Green, Blue and Red. The green and the blue colors are associated to Rayleigh distribution with higher p-values representing the very good and good regions while the red color represents the bad quality data with lower p-value.

3. EXPERIMENTAL RESULTS AND DISCUSSION

In this section we present the experimental results of the proposed approach. All the experimental tests are performed on sonar images acquired using DAURADE AUV robot. The sonar images are processed based on intensity data, where the darker gray color means low intensity of backscattering and the brighter color means high intensity of back scattering.

Fig. 1(a) is an example of raw sonar image acquired by DAURADE AUV, Fig. 1(b) represents the intensity correction of the raw sonar image using grey level normalisation process. In the Fig. 1(c), Fourier transform of the corrected sonar image is presented. The FFT image is used for the step of spectral filtering and residual extraction using Butterworth filter. Fig. 1(d) represents probability density function (PDF) of Fig. 1(c) for each 40 pixels of the range of the image. Fig. 2(a) is an example of filtered sonar image using Lee filter (standard deviation =9). Fig. 2(b) represents the PDF of Lee filtered image versus range image. In the Fig. 2(c), residual of Lee filtered image (residual=corrected sonar image – Lee filtered image) is presented. Fig. 2(d) represents the probability density function of the residual Lee filtered image versus range sonar. Fig. 2(e) represents the corrected sonar image filtered using Butterworth filter, Fig. 2(f) represents PDF of Butterworth filtered image versus range image. In the Fig. 2(g), the residual of Butterworth filtered image is shown and Fig. 2(h) the probability density function of the residual Butterworth filtered image versus range sonar is given. It can be observed that the
PDF in Fig. 2(h) is not constant and varies at far range of the sonar data, thus justifies the proposed approach by analysing the residual obtained through spectral filtering.

In order to show the quality mapping, we demonstrate the proposed approach on the famous sonar image of ‘Swansea’ acquired by DAURADE AUV given in Fig. 3. Fig. 4 represents a zoomed region of Fig. 3 given in rectangular red block. It can be observed from these two figures that the noise level varies from the near range to the far range of the sonar data. According to the proposed approach, the sonar image is partitioned into 64x64 size windows and transformed into frequency domain using Fourier transform. A high-pass Butterworth filter of order $n = 4$ with cutoff frequency $r_0 = 35$ given by equation (1) is applied using equation (2). The residual image is obtained by taking the inverse Fourier transform.

Rayleigh distribution is fitted to the residual image and its parameters are estimated by using ML method. For the acceptance of the null hypothesis, two thresholds $\tau_1 = \frac{1.36}{\sqrt{64^2}} = 0.02121$ and $\tau_2 = \frac{1.22}{\sqrt{64^2}} = 0.019125$, are selected based on the significance level ($\alpha_1 = 0.05$ and $\alpha_2 = 0.10$) of the KS test for Gof [12].

Fig. 5(a) represent the cumulative distribution of the residual data along with CDF of the fitted Rayleigh distribution with p-value greater than $\tau_1$, Fig. 5(b) represent the CDF of the fitted Rayleigh distribution with p-value greater than $\tau_2$ but less than $\tau_1$, respectively. From Fig. 5, it can be observed that the fitted Rayleigh from the residual data and the fitted Non-Rayleigh distributions are disjoint.

Based on the two sample KS tests statistics, if the critical value p is greater than $\tau_1$, the null hypothesis is accepted (the data follows the Rayleigh distribution) with strong support ($\alpha_1 = 0.05$), which represent the good quality data being represented in green color blocks in the Fig. 6 and Fig. 7. Similarly, if the critical value p is greater than $\tau_2$ and less than $\tau_1$, again the null hypothesis that the data follows the Rayleigh distribution is accepted but with relatively less support ($\alpha_2 = 0.10$), which is represented by the blue regions in the Fig. 6 and Fig. 7. The regions which are not following the Rayleigh distribution represent those far regions where the signal response is very weak, represented by the red regions in the Fig. 6 and Fig. 7. In Fig. 8, we give an example of the portion of mapping along with the corresponding p values and color representation.

The results of the proposed mapping approach can be confirmed by the visual inspection of the raw sonar image and the mapped data. It has been observed that false positives (regions not following Rayleigh distribution) are also detected which limits the capabilities of the proposed approach in complex environments with high coefficient of variation.

4. CONCLUSION

Coherent imaging systems (like synthetic aperture radars, sonar, ultrasound and laser imaging) are commonly affected by multiplicative noise (also known as speckle noise). This article presents a novel statistical method for transforming the sonar data into quality mapping by exploiting the noise distribution in the sonar data. Based on the hypothesis that the noise follows the Rayleigh distribution, the proposed approach map the data into acceptable (good) region otherwise map the data into a non-acceptable (bad) regions. In the future, further analysis about the noise distributions in complex environments would make the proposed quality map more accurate and precise.
Fig. 1: a) Example of raw sonar image, b) Corrected intensity of the raw sonar image, c) Fourier Transform of the corrected sonar image, d) Probability density function (PDF) vs. range sonar.

Fig. 2: a) Corrected sonar image filtered using Lee filter, b) PDF of Lee filtered image vs range image, c) Residual of Lee filtered image, d) Probability density function of the residual Lee filtered image vs. range sonar.

e) Corrected sonar image filtered using Butterworth filter, f) PDF of Butterworth filtered image vs range image, g) Residual of Butterworth filtered image, h) Probability density function of the residual Butterworth filtered image vs. range sonar.
Fig. 3: Swansea Image

Fig. 4: Zoom region of the starboard in Fig. 1

Fig. 5(a): CDF of the Residual data with Rayleigh \((p > \tau_1)\) and Non-Rayleigh distribution

Fig. 5(b): CDF of the Residual data with Rayleigh \((\tau_2 < p < \tau_3)\) and Non-Rayleigh distribution

Fig. 6: Quality Mapping of the Swansea Image in Fig. 1

Fig. 7: Quality Mapping of the Image in Fig. 2

Fig. 8: Quality mapping with the critical value \(p\)
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REAL-TIME ACOUSTIC MONITORING OF BALEEN WHALES FROM AUTONOMOUS PLATFORMS

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Abstract: In the past decade, much progress has been made in real-time passive acoustic monitoring of marine mammal occurrence and distribution from autonomous platforms (e.g., gliders, floats, buoys), but few systems are capable of detecting the calls of multiple species simultaneously. We have combined the low-frequency detection and classification system (LFDCS; Baumgartner and Mussoline, 2011, JASA 129:2889-2902) with the digital acoustic monitoring (DMON) instrument to detect, classify, and report in near real time the calls of several baleen whale species, including fin, humpback, sei, bowhead, and North Atlantic right whales. The DMON/LFDCS has been integrated into the Slocum glider and APEX profiling float, and we have integration projects currently underway for the Liquid Robotics wave glider and a moored buoy. In a recent evaluation study, we deployed two DMON/LFDCS-equipped Slocum gliders in the central Gulf of Maine for 3 weeks during late November and early December 2012. The gliders reported over 25,000 acoustic detections attributed to fin, humpback, sei, and North Atlantic right whales. Real-time detections were evaluated after recovery of the gliders by (1) comparing the acoustic detections to continuously archived audio recorded by the DMON/LFDCS, and (2) comparing species-specific detection locations with nearby sightings collected from both an aircraft and ship. The overall false detection rate for individual calls was 14%, and for right, humpback, and fin whales, false predictions of occurrence during 15-minute
Agreement between acoustic detections and visual sightings from aerial and shipboard surveys was excellent (9 of 10 visual detections were accompanied by real-time acoustic detections of the same species by a nearby glider). We envision that this autonomous acoustic monitoring system will be a useful tool for both marine mammal research and mitigation applications.

**Keywords:** whale, glider, buoy, float, detection, classification, marine mammal, real-time

**Introduction**

Marine mammals are an integral part of the ocean ecosystem, but like most marine organisms, their occurrence, distribution, and abundance are a challenge to monitor from ocean observing systems. Detection of marine mammals has traditionally been accomplished by human observers during visual surveys. These surveys rely on the fact that marine mammals breathe air, and therefore must come to the surface periodically where they can be detected visually. This approach is exceedingly expensive, as it requires a large team of observers (typically 3-8 people) and a ship or aircraft. Moreover, visual surveys are limited by weather and sighting conditions, such as fog, rain, heavy seas, and darkness. For their expense, visual surveys are often inefficient, and with reductions in platform availability due to budget restrictions, they are becoming increasingly impractical.

In recent decades, passive acoustic recorders have become extremely popular for detecting vocally active marine mammals, as they can operate continuously for periods of months to years. Widespread use of passive acoustic monitoring faces two challenges: (1) passive acoustic recordings are only available for analysis after instruments are recovered; there is currently no real-time access to these data, and (2) analysis of passive acoustic recordings is slow and tedious, involving trained human analysts that pore over large volumes of acoustic data for months to assess occurrence and/or calling rates. In some cases, the delays in access and analysis are perfectly acceptable, but for mitigation applications, those involving real-time response (e.g., facilitating at-sea research or directing surveys), or where storage or recovery of audio recordings is not feasible, most passive acoustic recorders are unhelpful. There is an urgent need for real-time information on the occurrence of marine mammals for both science and mitigation applications.

**Materials and Methods**

Scientists and engineers at the Woods Hole Oceanographic Institution (WHOI) have developed a system to detect and classify the calls of marine mammals in-situ and report those detections in near real time via Iridium satellite. The system relies on two enabling technologies: the digital acoustic monitoring (DMON) instrument and the low-frequency detection and classification system (LFDCS). The DMON hardware and LFDCS software (described below) currently provide the capability to (1) simultaneously detect and classify the calls of 4 species of baleen whales from Slocum electric gliders (Teledyne Webb Research), (2) transmit summary information (tallies) of all classified calls as well as detailed characteristics of a subset of individual calls to a shore side server via Iridium.
satellite, and (3) immediately post all transmitted data to a publicly accessible website [1]. We have also developed this same capability in APEX profiling floats (Teledyne Webb Research), and we have projects in progress now to integrate this detection and classification capability with two additional autonomous platforms: moored buoys and Liquid Robotics, Inc. (LRI) wave gliders (Fig. 1).

![Autonomous platforms for monitoring baleen whale occurrence and distribution.](image)

**DMON**

The DMON [2] is a hardware device that can sample from up to three externally mounted hydrophones, and uses a pre-amplifier and anti-alias filter to condition the audio from each of these channels before digitizing with a 16-bit A-D converter. Three hydrophones are available for use with the instrument covering low- (8-7500 Hz), mid- (0.1-50 kHz), and high-frequency (1-160 kHz) bands. The DMON features a programmable Texas Instruments TMS320C55 digital signal processor (DSP), 32 GB of flash memory, and serial (RS232/485) input/output lines. The instrument is extremely low power, making it ideal for use on power-limited autonomous platforms; when running the LFDCS software, the instrument consumes just 130 mW of power. The design of the DMON is “platform agnostic,” meaning that the DMON needs to know very little about the platform upon which it is deployed; this makes integration into new autonomous platforms straightforward.

**LFDCS**

The DMON is analogous to a smartphone in that the DMON (a hardware device) can run applications (software) that implement specific detection and classification algorithms. Baumgartner and Mussoline [3] developed the LFDCS for identifying baleen whale calls based on pitch tracking and quadratic discriminant function analysis, and we have converted the LFDCS into a DMON application. The LFDCS algorithm uses dynamic programming to estimate a pitch track for any type of narrowband call (Fig. 2). A pitch track is a compact representation of a sound (analogous to a series of notes on a page of sheet music) derived from an audio spectrogram; it consists of a time series of frequency-amplitude pairs that describe the frequency and amplitude modulation of a sound.
Attributes of the pitch track (e.g., start frequency, end frequency, duration, slope of frequency variation) are extracted and compared to the attributes of known call types in a call library using quadratic discriminant function analysis. The call library can contain hundreds of these known call types, thus allowing the LFDCS to efficiently detect and classify many different calls produced by numerous species. Baumgartner and Mussoline [3] compared the performance of the LFDCS to that of several human analysts for low-frequency sei whale (*Balaenoptera borealis*) downsweeps and right whale (*Eubalaena glacialis*) upcalls, and found that the accuracy of the LFDCS was similar to that of an analyst. In addition to right whale upcalls and sei whale downsweeps, Baumgartner et al. [1] found that the LFDCS performs well for fin whale (*Balaenoptera physalus*) 20-Hz pulses and several types of humpback whale (*Megaptera novaeangilae*) tonal calls as well. Feasibility tests have indicated that the LFDCS is also effective at detecting the narrowband calls of bowhead whales (*Balaena mysticetus*), blue whales (*Balaenoptera musculus*), and several species of ice seals, and we are currently expanding the system to include detection of odontocete whistles.

during operation aboard an autonomous platform, the DMON running the LFDCS application (referred to hereafter as the DMON/LFDCS) relays three types of information to the platform computer: (1) detailed detection data, including pitch track and classification information, for every sound detected, but only up to a maximum of 8 kB of detection data per hour, (2) summary detection data for each call type in the call library, consisting of tallies of classified sounds (relayed every 15 minutes), and (3) instantaneous background noise spectrum (relayed every 30 minutes). The autonomous platform is

**Fig. 2:** Pitch tracks and associated classification information received in real time from a Slocum glider in the central Gulf of Maine on (a) December 2, 2012, and (d) November 17, 2012. Pitch tracks are shaded according to amplitude relative to background; dark gray shades are high amplitude, light gray shades are low amplitude. For classified calls, the species producing the call is shown above the plot and the Mahalanobis distance is shown just below the species abbreviation: FW for fin whale 20-Hz calls, and RW for right whale up calls. (b,c) Spectrogram of archived audio corresponding to the same time period as that shown in (a). (e) Spectrogram of archived audio corresponding to the same time period as that shown in (d). The spectrogram in (b) shows a humpback whale song, while the corresponding low-frequency spectrogram in (c) shows repeated fin whale 20-Hz pulses. Broadband sounds in (e) were produced by the glider rudder.
responsible for storing these data and transmitting them to a shore-side computer via an Iridium satellite modem every 3 hours (Fig. 1).

Summary detection data consist of tallies of every call of a particular call type detected during the tally period (i.e., the 15-minute period between summary detection reports). It is with these data that the occurrence of a species is predicted. Predictions are based simply on the number of calls tallied during the tally period for a particular species; if the number of calls (N) is equal to or greater than a predetermined minimum call count threshold (N\textsubscript{min}), there is a 95% or greater probability that the species is present if it is vocally active. The threshold for a particular species is determined by comparing analyst-detected acoustic occurrence (presence/absence) of the species in DMON/LFDCS audio recordings to the number of tallied calls for that species reported in real time by the DMON/LFDCS using logistic regression (after [1]).

In addition to detecting, classifying, and reporting whale calls, the DMON/LFDCS also records audio to flash memory. Audio from the low-frequency hydrophone is currently sampled at 60 kHz, low-pass filtered, decimated to 2-10 kHz (user selectable), compressed using a lossless algorithm [4], and saved in the DMON’s 32 GB of flash memory. For long-term applications, duty cycling is used to record only short periods of audio to avoid overflowing the limited memory (e.g., 15 minutes of audio every hour). These recordings can be accessed upon recovery of the autonomous platform and post-processed to verify the accuracy of archived real-time detections.

Results

We recently conducted a pilot field project during late November and early December 2012 during which two Slocum gliders equipped with DMON/LFDCS were deployed in the central Gulf of Maine (described in [1]). The gliders remained in the study area for 3 weeks reporting real-time acoustic detection and classification data via Iridium satellite every 2 hours; these data were posted in real time to the website dcs.whoi.edu (these data are still accessible there). During the last week of their mission, observers aboard the 56-m R/V Endeavor rendezvoused with the gliders to validate the real-time acoustic detections. During 9 instances when shipboard observers located whales, and a glider was within 20 km of the whale location, the glider correctly predicted the occurrence of the sighted species within ±12 hours of the sighting time in 8 (89%) of the 9 cases [1].

After recovery of the vehicles, the real-time detections were compared to audio recorded by the DMON/LFDCS. The occurrence of calls detected by a human analyst was highly correlated with the number of calls detected by the DMON/LFDCS during the 15-minute tally periods (logistic regression; p < 0.01 for all species). The occurrence of fin, right, and humpback whales were falsely predicted in 5% or less of the 15-minute tally periods, while the missed occurrence rate was 50% for fin and right whales and 81% for humpback whales. Missed occurrence rates were moderate to high because the LFDCS cannot both detect and pitch track very quiet calls (< 11 dB above background); however, for mitigation applications, falsely identifying occurrence is more egregious than missing calls, particularly for species that call often enough to provide multiple opportunities to correctly identify calls. These comparisons and performance statistics indicate that when appropriate thresholds are applied to the detection data, confidence in the resulting DMON/LFDCS predictions of occurrence is high.

In addition to reporting summary detection data, the DMON/LFDCS also transmitted to shore detailed detection data (i.e., pitch tracks) for 25% of all detected sounds, allowing assessment by a human analyst in real time. Review of these pitch tracks allowed the
unambiguous identification of humpback and fin whale song as easily recognizable patterns of frequency-modulated calls or repeated 20-Hz pulses, respectively (Fig. 2). Although the performance statistics cited above do not account for this analyst review, the availability of pitch tracks in real time allowed an analyst to determine with nearly 100% certainty the occurrence of humpback and fin whales independent of the onboard classifications.

**Conclusion**

A real-time acoustic detection, classification and reporting capability has numerous applications for both science and conservation, and we envision that real-time passive acoustics from autonomous platforms will become a routine tool for marine mammal and ocean noise monitoring. Such a real-time capability can improve the efficiency of traditional visual survey methods by identifying areas where animals are likely to be located, and can provide critical occurrence information in sensitive areas where human activities must be managed to avoid harmful interactions with marine mammals.

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SONOBOT - AN AUTONOMOUS UNMANNED SURFACE VEHICLE FOR HYDROGRAPHIC SURVEYS WITH HYDROACOUSTIC COMMUNICATION AND POSITIONING FOR UNDERWATER ACOUSTIC SURVEILLANCE AND MONITORING


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Abstract: Paper describes the autonomous surface vehicle Sonobot – a lightweight surface vehicle for shallow water hydrographic surveys, research, monitoring or surveillance. Capable of both autonomous and remotely controlled operation, this fast and maneuverable unmanned vehicle represents a cost-effective solution for surveys in harbors, inland and coastal waters, hard to reach or dangerous locations. Design of the vehicle is discussed in the paper, followed by an overview of the platform and practical results of several survey missions.

Keywords: Autonomous, mobile robots, data acquisition, monitoring, sensors, ultrasonic transducers, positioning systems
1. INTRODUCTION

Unmanned surface vehicles (USVs), also known as autonomous surface craft (ASC), were first developed and used in naval operations as early as after the World War II for purposes such as minesweeping and battle damage assessment [1].

USVs evolved over decades, becoming more effective and affordable, and with the modern benefits of wireless data transmission technologies and precise global positioning systems, the USVs are becoming more and more attractive for a wide range of commercial, scientific and military operations.

For modern unmanned vehicles, advanced radio and satellite communication technologies enable live feedback from on-board sensors, remote control or complete autonomy. Further USV research and development efforts focus on greater communication ranges, more autonomous functionality, easier and more reliable launch and recovery, greater sensor performance, better power management and longer operation times.

The USVs are presented in various hull and craft types, such as semi-submersible craft, conventional planing hull craft, semi-planing hull craft, hydrofoils, catamarans and many others. With sizes ranging from small vehicles to large unmanned boats, more and more prototypes and commercial USV solutions are becoming available on the market.

Although there are still significantly fewer USVs than their unmanned underwater vehicle counterparts (UUV), demand for USVs grows for many applications.

USVs for naval operations are often based upon traditional surface vessels, as controls, navigation and telemetry systems can transform almost any conventional craft into a USV, remotely operated by crew ashore or on other vessels. Although unmanned, these systems depend on telemetry and are more similar to underwater remotely operated vehicles (ROVs) than autonomous underwater vehicles (AUVs) [2].

A greater variety of hull forms was developed for academic and commercial survey purposes. Survey applications trend toward small, low-cost USV platforms that can be used for a variety of missions [3].

Smaller USVs are particularly attractive for hydrographic surveys aimed at measuring the depth and bottom configuration of water bodies, as an autonomous or remotely controlled surface vehicle can be equipped with radio- and underwater acoustic communication and positioning devices for underwater acoustic surveillance and monitoring, as well as with on-board echo-sounder, side-scan sonar and other sensors to collect valuable data in automated mode. As geological mapping of the seafloor in nearshore or shallow water areas presents technological challenges due to the dynamics of the environment, high volume of data collected, and the limitations of operating in very shallow water [4], a remotely operated or autonomous seafloor mapping USV is a great solution for the problems of shallow water settings.

The Sonobot, an autonomous USV from Evologics GmbH, Germany, was designed and built to address the industry need for a light and affordable platform for hydrographic surveys, mobile radio- and underwater acoustic communication with underwater assets and their positioning for underwater acoustic surveillance and monitoring, as well as other missions. We will further discuss the design of the USV Sonobot vehicle, overview the platform and provide a few examples of its applications.
2. USV SONOBOT DESIGN CONSIDERATIONS

EvoLogics GmbH, Germany, designed and built the USV Sonobot to meet the need for an effective low-cost solution for hydrographic surveys and other missions in shallow inland and harbor waters, as well as in adjacent coastal waters.

The main idea behind the USV Sonobot was to present a small, simple and usable platform for planning and executing a hydrographic survey, underwater positioning of cooperating mobile and stationary underwater assets, as well as mobile radio- and underwater acoustic communication between a surface station and mobile/stationary underwater assets. The design of the system ought to deliver accurate geo-referenced bathymetric and positioning measurements, high-data rate communication and high-quality imagery with minimum transport, launch and recovery efforts.

The concept is to offer the end user a modular system with several configuration options and deliver a ready-to-use USV with all the on-board equipment installed and ready for immediate action.

The platform, a twin-hull catamaran craft, was chosen for its great stability and payload capacity.

After initial trials with a laser rangefinder, the USV Sonobot was equipped with a highly accurate differential (/RTK) GPS system to yield an exact location of each depth measurement and/or underwater asset’s position estimation.

Specifically for the vehicle, the patented S2C (Sweep Spread Carrier) broadband communication technology was used to build an advanced single-beam echo sounder, capable of delivering precise and accurate depth measurements even in very shallow waters.

As an option, an S2C technology-based underwater acoustic communication and positioning system was integrated into the vehicle. The system provides full-duplex data exchange, offering many data management options along with addressing and networking with multiple mobile underwater vehicles and stationary seafloor instruments, as well as tracking and navigation of underwater assets. Both features of the system complement each other in a fully integrated system: positioning data is calculated simultaneously with acoustic transmissions, so there is no need for switching between positioning and communication modes.

The side-scan sonar, GPS system and other equipment options were selected among commercial off-the-shelf products to best fit the USV Sonobot platform and offer the user the optimal configuration for his particular requirements.

As the USV was designed to be convenient for day-to-day use by surveyors, service providers or scientists, essential features for easy launch, operation and recovery were developed. In particular, the vehicle was constructed of lightweight materials and was equipped with smart fittings for simple and fast assembly without any tools or special skills. Carbon fiber was chosen for the hulls to add robustness and make the vehicle suitable for operations in harbor seawater or waters polluted with industrial waste.

To execute fully-automated survey missions, the vehicle was fitted with an autopilot enabling waypoint-based navigation. It was equipped with a radio control unit, enabling either manual operation or supervised autonomy mode, where the operator can interrupt the autopilot and remotely steer the vehicle.

Introducing a front-view camera allowed for better control in remotely operated mode and added surveillance functionality. The USV can thus operate in hardly accessible areas with no direct visibility from the shore.
A wireless LAN station on the shore was included for on-demand or real-time data acquisition during the mission. A pre-configured PC was added to the on-shore setup to monitor the system during the mission and preview the bathymetry data collected.

Too large payload capacity was given up for better transportability, so the USV Sonobot system fits into a car trunk and does not require a bulky trailer or container. Total system weight of the vehicle was kept as low as possible, so the resulting vehicle can be transported and lifted by one person.

A detailed system overview follows in the section below. A few examples of its applications then round up system description.

![Fig. 1. The USV Sonobot](image)

3. SYSTEM OVERVIEW

In this section, we will take a closer look on each of the USV’s components: the vehicle platform, the on-board equipment and the shore equipment comprising the USV Sonobot system.

The vehicle consists of the main body with the on-board equipment, mounted on crossbar supports that attach to the twin hulls, housing the batteries and integrated hydrojet thrusters. It is 1.2 m long, 0.92 m wide, 0.5 m high and weights up to 30 kg depending on system configuration.

The vehicle was designed for an effortless no-tool assembly, therefore the main body, the crossbars and the floaters are held together by eccentric latches that are both secure and fast to unlock.

The on-board equipment of the USV includes: a differential GPS system, a single-beam echo-sounder, a side-scan sonar, a wireless communication system, a radio control system, an autopilot and the on-board PC, and optionally also an underwater acoustic communication and positioning system.

The shore equipment comprises a wireless LAN tripod station, a remote PC and a radio control unit.
**Propulsion.** Each floater-hull made of carbon fibre houses a 43 mm hydro-jet thruster powered by a 700 W brushless electric motor, creating a total thrust of at least 100 N. There are two options for the motors used in the vehicle: motors from Lehener Motorentechnik, Germany, or motors from Kontron, Germany. The first motor option has a gear drive. Second option has a direct (gearless) drive. A special scheme for tracking the current height was developed and implemented to control and safely and reliably operate the motors.

The motors can drive in reverse, enabling precise steering and maneuvering. With its two 700 W jet thrusters, the vehicle can speed up to 13 km/h (7 kn), but for survey purposes is best to operate at 4 km/h (2.2 kn) to keep the balance between measurement accuracy and propulsion speed.

The catamaran-type build of the USV Sonobot ensures it can be operated under up to 3 Bft (3.4 - 5.4 m/s) wind and 0.3 m wave conditions, and the system is thus well suitable for missions in most inland water bodies, sea harbours and adjacent coastal waters.

**Power management.** The standard USV’s propulsion system is powered by two sets of 14.8V Lithium Polymer or NiMH rechargeable batteries. The batteries are held in the hulls of the vehicle and yield a total capacity of up to 80 Ah. The payload is powered by two 14.8V 10Ah battery packs inside the vehicle’s main body. Alternatively, the propulsion system can be powered by two sets of Lithium Polymer rechargeable batteries held in the hulls of the vehicle, yielding a maximum total capacity of up to 168 Ah.

Charging the batteries is performed with a battery charger simply plugged into the back of the USV when returned to the shore.

A fully charged system’s operating time is up to 20 hours at optimal speed (4 km/h), corresponding to an operating range of 80 km. During the mission, the battery status can be monitored on the on-shore PC screen.

**DGPS-System.** A differential GPS (DGPS) system with improved location accuracy is installed on the vehicle to enable precise positioning of the USV.

The positioning system consists of the following components:
- JAVAD GPS Antenna, type: Grant-G3T,
- JAVAD GPS Receiver, type: DELTA TRE-G3T (GPS, GLONASS, Galileo),
- ALLSAT GSM/GPRS modem, type: come2ascos (RTK corrections).

The JAVAD DGPS antenna with access to GPS, GLONASS, Galileo and ASCOS is coupled with a vertical reference unit, enhancing the location accuracy of the echo-sounder’s acoustic transducer to the sub-centimetre range. Hence, the water level deviations can be evaluated during the mission.

**S2C - Echo-Sounder.** The integrated EvoLogics echo-sounder is based on the EvoLogics patented S2C technology. It provides accurate depth measurements ranging within 0.5 – 60 m or optionally 0.5 – 100 m.

Unlike conventional echo-sounders, the depth measurements are carried out with broadband (spread frequency) pulses with subsequent complex signal analysis. The echo-sounder operates at 80-120 kHz and measures the depth with 6 mm accuracy. Therefore, with 2 measurements per second the USV, moving at 4 km/h, can collect enough data for a very detailed profile of depth and bottom configuration of the water body.

For significant drug reduction during the movement, the echo-sounder’s transducer is encapsulated into a streamlined, hydro-dynamically optimised housing.

**Side-Scan-Sonar.** The side-scan sonar for geo-referenced seafloor imaging is installed into each of two carbon fibre floater-hulls to aid bottom observations and find and identify objects in the water.
It has a vertical opening angle of 60° and operates at 680 kHz and 340 kHz. Depending on the frequency, the operating range is up to 100 or 200 m in both directions. An important feature of the side-scan sonar is the use of broadband sweep signals facilitating high-resolution measurements of between 1.0 and 1.5 cm (depending on the frequency).

**S2C underwater communication and positioning.** As an option, a combined S2C M-series underwater acoustic communication and positioning system of light and compact design enables trans-media access to underwater assets, as well as their positioning (navigation/telemetric operation) during the mission both in local and geographical coordinates. The system provides full-duplex data exchange and supports many data management options along with addressing and networking with multiple mobile underwater vehicles and stationary seafloor instruments. Simultaneously, it enables tracking and navigation of underwater mobile assets with no need for switching between positioning and communication modes.

For different operation ranges and application scenarios, the transceiver of the communication and positioning system can be either omnidirectional or have directivity in vertical and/or horizontal plane (with an opening angle of 70-110 degrees). Depending on the model, the operation ranges of communication and positioning system lie between 1 and 6 km.

The transceiver of the system is fitted into a streamlined, hydro-dynamically optimised housing, minimising drag reduction even when vehicle moves fast.

**Autopilot.** The autopilot enables fully autonomous operation of the vehicle and consists of a navigation and control system, installed on the vehicle, and the software for mission planning, installed on the shore control PC.

A mission plan can be created on any geo-referenced map with set of waypoints that define a survey grid that can be configured, saved and loaded for a repeated run of the same route. The on-shore PC software allows to monitor the mission in real-time and use the radio control to overrun the autopilot and steer the vehicle manually.

**Radio control.** The vehicle can be remotely controlled over a digital 6-channel 2.4 GHz radio system utilizing the Robbe Spectrum DX6i technology. The remote control pad allows to precisely steer and manoeuvre the USV within up to 2000 m range. To provide this large operating distance, the radio control system was significantly modified with custom-designed boosters. To enable operation in harsh weather conditions, the control pad of the radio control was mechanically modified with supplementary waterproof elements.

The vehicle is equipped with a front-view camera to aid the manual steering, but it can be used for a monitoring or surveillance mission in hard-to-reach areas.

**Wireless LAN-Station.** A wireless LAN network (based on WLAN antennas from L-com and access points from ACKSYS or Bullet) with a tripod base station on shore connects the embedded on-board PC with the shore PC with up to 108 Mbps speed. An omnidirectional wireless antenna covers 2 km, which for many missions is within the surveyed area, enabling uninterrupted real-time access for monitoring or data acquisition during the mission (e.g. on-line collection of geo-referenced pictures taken by the vehicle and geo-referenced bathymetry images from vehicle measurements). A directional WLAN antenna can extend the communication range of the system if needed.

**On-board PC.** An embedded processor with an integrated 2.5” 128 GB SSD (Solid State Drive) logs all raw measurement data in ASCII format, compatible with multiple GIS, CAD and spreadsheet software products. Through wireless LAN, the data is accessible from the shore and can be retrieved during the mission.
As the specialized DUNE Uniform Navigational Environment software, developed by FEUP, Porto University, was integrated into the USV’s hardware platform, the vehicle is able to inter-operate with diverse systems, which interfaces match the requirements of DUNE, in particular, with multiple underwater and surface vehicles and stationary seafloor assets.

**Shore control PC.** A shore control PC provides remote access to all mission data through a wireless link to the embedded PC on the vehicle.

The shore control software, including the mission planner and the system monitoring tools, is pre-installed on the PC and provides the information about the current mission status, the battery voltage, the acoustic signal quality etc. The software allows to modify the settings all the vehicle’s onboard instruments before or during the mission.

The on-shore PC with Drop/Shock/Spill protection and anti-glare screen is robust and suitable for field operations.

Optionally, when the underwater acoustic communication and positioning system is installed on the vehicle, an integrated version of the GUI can be used for remote guidance, control and monitoring of the underwater mobile and stationary assets, as well as for obtaining accurate geo-referenced underwater images (bathymetry, photos, parameters of the environment, etc.) from all data sources currently accessible over the acoustic network.

4. APPLICATION EXAMPLES

The USV Sonobot was extensively used in various lakes and rivers both in Germany and abroad. A typical application can be illustrated with a bathymetric survey, conducted at a flooded opencast mine - now the Karlssee lake near the town of Förderstedt, Germany.

The lake, about 650 m long and 200 m wide, is surrounded by vegetation and thus hardly accessible with bulky equipment, as well seen in both the pictures taken at the shore and the satellite image of the area (Fig. 2a,b).

![Fig. 2. Karlssee lake bathymetry: a satellite image of the location (a), and Karlssee bathymetry: the results in 3D (b)](image)
Using a manned boat or a larger USV for the survey would require both time and personnel to perform the launch and recovery. The vehicle and all accessories were transported to the site in a pick-up trunk and was quickly set up for the mission. On autopilot, it followed the mission’s waypoints and performed all the bathymetric measurements as planned. The precise 3-dimensional bathymetric map of the lake bottom is presented in the Fig.2b below.

A side-scan sonar imaging of a river bottom near a bridge is a good example of using the USV for monitoring and maintenance of underwater infrastructure. The side scan sonar images, layered over a satellite image of the area, is presented in the Fig.3a,b.

Fig. 3. Side-Scan Sonar images of a river bottom near Mühlendamm in Rathenow, Germany

Fig.4a shows the operator’s screen (the shore control PC with the mission planning software running. Waypoints given by the user define the survey grid for bathymetric scan of the area (Sacrow-Paretzer channel in Schlänitzsee lake). The autopilot enables moving the vehicle along the grid lines. The mission data saved on the vehicle is also transmitted to the on-shore PC over WLAN, allowing to monitor the mission in real-time.

Fig. 4. Control PC screen with waypoints defining a survey grid (a), XY plot of the surveyed area obtained during the mission in real time (b, below), 3D plot of the surveyed area estimated as a result of combining echo-sounder data and side-scan data (b, above)
An example of side-scan data being in real time received on the on-shore PC is shown in Fig.4b below. After completing the mission or overrunning the autopilot (e.g. over radio control manually interrupting the mission), the on-shore PC software uses the combination of echo-sounder data and side-scan data for estimating the 3D plot of the surveyed area. The 3D plot of the bathymetry obtained in the area is demonstrated in Fig.4b above.

Apart from the functionality offered by the manufacturer, users of the USV Sonobot are provided with instruments to develop their own customised functions. Being an open-source software, the DUNE Uniform Navigational Environment provides an opportunity to replace and/or integrate additional sensors, replace and/or integrate individual software components running onboard the vehicle and/or the coastal station. To facilitate such user-specific development and customization of the vehicle, extended sets of service data can be configured for on-line transmission from the vehicle to the coastal station during the mission.

For example, Fig.5 and Fig.6 visualize several data streams for detailed investigation of the vehicle’s own autopilot functionality. Fig.5a and Fig.6a demonstrate the performance of the autopilot as the vehicle operates in a river with a current of about 2 knots. At scans along Y-axis, the river flow was transversal to the vehicle route (as the river flows along the X-axis). The difference between Fig.5a and Fig.6a is the general direction of the vehicle movement, in first case the direction is down the river flow, in the second case – up the river flow.

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**Fig. 5. Example: investigation of the autopilot functionality (scans proceed downwards the river flow)**

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**Fig. 6. Example: investigation of the autopilot functionality (scans proceed upwards the river flow)**
Related to these figures of the vehicle paths, detailed data plots are presented in Fig. 5b and Fig. 6b, that demonstrate desired headings and control headings as the vehicle moved along the route. Apart from that, Fig. 5c and Fig. 6c show the rotation angles of the vehicle during its mission for custom post-processing of the side-scan sonar data obtained during the mission.

With this data on hand, the user, for example, has an opportunity to develop his own, application-specific autopilot algorithms. Similarly, other data, output from all vehicle sensors (including customer-specific ones) can be transmitted to the coastal station, thus treating the USV Sonobot system as an expandable robotic platform and so enabling an opportunity for further developments of system functionalities.

CONCLUSION

An effective turnkey solution, the vehicle delivers precise and accurate bathymetry data in autonomous mode for surveyors, service providers and scientists, who can benefit from detailed information about the depth and floor structure of water bodies. The USV Sonobot can cover 100-200 hectares in one mission and does not require more than one operator to transport, launch and retrieve it after the mission.

The USV Sonobot represents a useful platform not only for hydrographic surveys in inland waters, harbours and adjacent coastal waters, but also, owing to its trans-media (radio and underwater acoustic) communication and positioning capabilities, for autonomous and remotely controlled missions designed to support operation of underwater mobile assets and interaction with remote seafloor instruments.

The system offers numerous options for the vehicle configuration and development of new functionality to expand the application range of the vehicle.

REFERENCES

THE PERSISTENT MARITIME MONITORING SYSTEM (PMMS)

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Abstract: This paper presents a system of systems concept for affordable and persistent surveillance and monitoring of large ocean areas by a self-deploying system of surface, submerged, and bottom-mounted sensor nodes. We begin with a brief review of maritime security and fisheries management missions that provide a technical, operational, and economic context for the system. We then describe the system concept in detail, including the USV and potential bottom-mounted and submerged nodes, and the seafloor-to-shore communications architecture. Next, we discuss four key enabling technologies that we have developed, both internally and with partners: the persistent USV platform (“Wave Glider SV3”), the seafloor to shore communications payload (“Gateway”), a suite of maritime surveillance and monitoring payloads (“Sentinel”), and a centralized autonomy system for coordinating and managing the control of large networks of vehicles at the fleet scale (“MVCA”). For each of these technologies, we provide technical details of the existing capabilities, review performance results from sea trials and operational deployments, and discuss our technology roadmaps for further development. Taken together, these mature and developing technologies and operational capabilities validate our concept for persistent and economical acoustic, RF, and imaging surveillance and monitoring of large areas of the ocean with a self-deploying, unmanned system of systems.

Keywords: USV, Wave Glider, acoustic communications, passive acoustic monitoring

MOTIVATION: MARINE FISHERIES SECURITY MARKET

The combined gross domestic products (GDPs) of all the world’s countries totalled approximately $72T in 2012. Of this vast figure, food production is by far the largest single industry, accounting for almost 6% of the total. Fisheries and aquaculture, and associated down-stream industries, in turn account for 10% of the food production total, or approximately $430B annually in 2012. Many nations, both developed and developing,
rely upon their fishing industries to generate export revenues and to provide jobs for large percentages of their populations. Marine fisheries are estimated to employ 260 million people globally, including fishing, processing, packaging, and marketing of the finished products [1]. Fishing is not only big business, but also an important component of food security at local, regional, and global scales. The Food and Agriculture Organization (FAO) of the United Nations estimated that marine animals (fish, shellfish, etc.) accounted for 16.7% of the global population's total consumption of animal proteins in 2010, providing 4.3 billion people with 15% or more of their dietary animal protein intake [2].

In all nations with significant ocean access, the renewable natural fisheries resources present in their exclusive economic zones (EEZs) and marine protected areas (MPAs) are valuable economic assets. For these nations, effective management and protection of these resources is vital to the health and livelihood of their citizens, and oftentimes to the national identity or cultural heritage of the nation as a whole. To provide effective husbandry of these resources, scientists and government organizations need to collect data on the type, location, quantity, and migratory patterns of fish throughout the ocean. This information is used to estimate fish stocks, monitor endangered species, determine the timing and duration of fishing seasons, to establish catch limits for commercial fishermen, and to monitor the overall health of the aquatic ecosystem.

Cultural and economic value notwithstanding, even wealthy nations have difficulty maintaining effective monitoring and control over their national fisheries due to the sheer size (and possible remoteness) of their EEZs and MPAs. For example, the United States has established the Papahānaumokuākea Marine National Monument, a 1200-mile-long area surrounding the Northwest Hawaiian Islands that has been designated a marine protected area with Hawaiian cultural heritage significance. Currently, this MPA exclusion zone is monitored via infrequent (roughly monthly) over-flights by US Coast Guard P3 aircraft. If the P3 detects a fishing vessel inside the exclusion zone, a USCG cutter may be dispatched from Honolulu to attempt an interdiction, arriving on station no sooner than 12 to 30 hours later. Meanwhile, the illegal fishing vessel, which will have noted the P3 fly over, will have ample time to recover their fishing gear and exit the exclusion zone well before the USCG cutter can arrive on station. Given the vastness and remoteness of the Papahānaumokuākea area, and even in the face of one of the world’s greatest maritime powers, illegal fishing vessels routinely operate in these waters with relative impunity [3].

This problem is compounded many fold for smaller and less wealthy nations. For instance, Palau is a small country in the western North Pacific Ocean with a population of only 21,000 and a total land area of just under 460 km². However, this land area is spread over more than 300 small islands of the Caroline Islands chain, resulting in a truly vast EEZ fishery that dwarfs the land area of the nation. Palau’s EEZ fishery is by far its greatest national economic asset, yet it too small – both in terms of population and economic prowess – to support a maritime law-enforcement infrastructure to deter illegal exploitation of its fisheries. In the absence of an effective deterrent, illegal fishing by poachers has become rampant in Palau’s EEZ waters, leading to significant over-fishing and threatening the collapse of the fishery as a whole. Furthermore, actors that are willing to engage in illegal fishing are also often engaged in other deleterious activities, such as human trafficking, abusive labour practices, unsafe working environments, and disregard for sanitation and food safety regulations. Taken all together, the negative economic impacts from illegal, unregulated, or under-reported (IUU) fishing activities can far exceed the lost value of the stolen fish. Palau’s IUU fishing problem is neither isolate nor uncommon. Agnew, et. al. [4], estimated that 20% of the world-wide marine fish catch is derived from illegal marine fishing, resulting in an annual direct loss of up to $23B, and even higher indirect costs.
PERSISTENT MARITIME MONITORING SYSTEM

Liquid Robotics has developed a system of systems concept – the Persistent Maritime Monitoring System (PMMS) – to address marine fisheries, commercial maritime security, and maritime defence markets with an affordable, scalable, and persistent ocean monitoring and surveillance network, Fig. 1. The two EEZ/MPA fisheries security examples discussed above, as well as several related marine infrastructure protection, marine mammals monitoring, and maritime domain awareness (MDA) scenarios that we have omitted for brevity, provide real-world mission context from which system-level functional and operational requirements can be derived. These scenarios also allow us to estimate market potential. The potential negative economic impacts embodied in these scenarios are significant and certainly sufficient to justify the costs of the system of systems that we describe below.

The Persistent Maritime Monitoring System is built around the Wave Glider™ SV3, which performs multiple roles as a long-endurance sensor platform, a seafloor-to-shore communications gateway, and a subsea navigation aid. As dictated by the needs of a particular mission or market, the PMMS can also incorporate unmanned aerial vehicles (UAS), unmanned underwater vehicles (both UUVs and buoyancy gliders), and unattended seafloor sensors to expand the coverage and enhance the sensing capabilities of the overall system. As a passive monitoring and queuing system, the PMMS will maintain constant contact with an on-shore or at-sea watch-floor, which will coordinate with patrol vessels or aircraft to perform interdiction operations.

The PMMS has three key advantages over traditional ship and aircraft based maritime monitoring approaches; it is persistent, scalable, and affordable. While it is conceivable to maintain continual monitoring with a fleet of manned assets, such operations would be prohibitively expensive in most applications. Even for wealthy nations with existing fleets of ships and aircraft, these costs are bearable only when they serve to protect strategic and high-value assets, such as the large group of ships, submarines, and aircraft that surround and protect aircraft carriers. In contrast with this “brute-force” approach, the PMMS achieves persistence at a relatively low and scalable cost that is commensurate with the inherent economic value of the protected assets (fisheries, marine mammals, marine

![Fig.1: Architectural overview of the Persistent Maritime Monitoring System.](image)
infrastructure, etc.). The PMMS is built upon long-endurance and all-weather unmanned vehicles and seafloor sensors, enabling uninterrupted ocean monitoring whether near shore or in remote reaches of the ocean. The PMMS is modular and scalable; the size of the field under surveillance grows linearly with the number of PMMS nodes employed, and the array of PMMS nodes can be tailored to provide varying levels of coverage and/or detection probability, in keeping with the nature of the monitoring mission. For instance, fisheries protection missions do not require dense coverage, since effective deterrence can be achieved if only even a few bad actors are detected and subsequently interdicted.

To realize our vision for the PMMS, Liquid Robotics has developed four key products and capabilities: (1) the Wave Glider SV3 USV, (2) an acoustic, RF, and SATCOM Communications Gateway module, (3) a suite of passive acoustic, EO/IR, and RF sensors for security missions, the Sentinel module, and (4) a multi-vehicle coordinated autonomy (MVCA) system to automatically organize and control the actions of large numbers of vehicles without imposing undue burdens of shore-based operators. These developments are presented in the following several sections, including an overview of design drivers, product status, and testing results.

**WAVE GLIDER SV3**

Released in the first quarter of 2014, the Wave Glider SV3 is the third-generation of Liquid Robotics’ popular line of unmanned surface vehicles, Fig. 2. Wave Gliders combine a unique wave-energy-harvesting propulsion mechanism with photovoltaic solar energy collection to achieve essentially unlimited endurance in open-ocean or near shore operations. The Wave Glider SV3’s hybrid surface/sub-surface architecture transfers heaving motions from surface waves down to a set of wings on the “glider” portion of the vehicle, Fig.2 (left). Since wave amplitude decreases exponentially with depth, these wings heave up and down through relatively quiescent water, thereby creating forward thrust. The reader is referred to the literature for more detailed discussions of the vehicle design [5,6] and of vehicle operational experiences in long-term deployments and in challenging environments [7-10].

*Fig. 2: The Wave Glider SV3 USV (left) and an exploded view of the surface float (right).*
The new SV3 vehicle design benefits from the experience we have gained through years of at-sea operations with the second-generation SV2 vehicles, and several technical advances that we have made in our wave propulsion mechanisms. We have also updated the vehicle’s control, communications, and sensing systems with the latest generation of commercially available electronics and components. The SV3 float is 38% (80 cm) longer than the SV2, but it collects twice as much solar energy per day, provides up to five times the peak payload power, increases the payload weight and volume capacity by 250%, doubles wave population speed in low sea states (SS1) and increases speed performance in higher seas (SS4) by 25%. It also includes line-of-sight and cellular data radios, and provides more than 100 times more onboard computational power and 1000 times more data storage capacity. We have also integrated several features into the SV3, such as a “pop-off” buoy with attached recovery line, to improve shipboard handling, launching, and recovery operations. Finally, the SV3 incorporates a high-efficiency electric thruster that can provide an episodic 1/2-kt speed boost (at 25W input power) to allow the SV3 to make headway against surface currents of up 2.5 kts and to maintain station in flat calm conditions. The thruster is integrated with the vehicle’s rudder module and allows vectored thrusting to spin the glider with respect to the surface float to eliminate twists in the umbilical that can occur under uncommon conditions.

COMMUNICATIONS GATEWAY

Liquid Robotics and its partners have integrated several different acoustic modem systems with the Wave Glider vehicles, including modems from Benthos, Evologics, LinkQuest, Sonardyne, and WHOI. Together with our partners, we have demonstrated reliable acoustic communications with seafloor sensors in deep water [12] by keeping station over the sensor, Fig. 3 (left), and as a mobile data retrieval system extracting stored data on a repeating cycle from a distributed network of seafloor sensors. The Wave Glider’s ability to precisely position itself directly above a seafloor sensor allows narrow transducer cones to be used in concert with lower power transmissions, thereby increasing the endurance of battery powered bottom nodes. Hydroid and WHOI have demonstrated the use of a Wave Glider for real-time acoustic communication with, and navigation aiding to an undersea vehicle while it conducted a hydrographic survey [13], Fig. 3 (right).

Once a subsea sensor or underwater vehicle’s data has been uploaded acoustically to the Wave Glider, the SV3 provides multiple communications channels to immediately relay sensor data to shore or local support vessels. In open ocean missions, data is relayed

![Image of Wave Glider and seafloor sensor communications](image)

**Fig. 3:** The Wave Glider serves as a multi-modal communications gateway between seafloor nodes, undersea vehicles, surface ships and platforms, aircraft, and shore facilities.
to shore via the Iridium satellite network. For near shore applications, a cellular data modem provides higher data rates at significantly lower costs. An RF data modem facilitates high-speed data exfiltration when operating in proximity (3 to 5 km, typically) to a support vessel or offshore marine platform. All three of these communications systems are standard on the SV3 generation of Wave Gliders, and on-board software allows the vehicle to automatically switch between communications channels to maximize data throughput and minimize transmission costs. Once vehicle data reaches Liquid Robotics’ shore-based command and control servers, it is archived and presented to appropriately credentialed and authenticated customers for secure real-time data streaming and/or for batch download.

WAVE GLIDER SENTINEL SYSTEM

The Wave Glider Sentinel system is a collection of payloads tailored to maritime monitoring and surveillance missions, and includes a passive acoustic sensor, a camera, an AIS receiver, and an optional RF direction-finding system. The passive acoustic sensor, developed jointly by Liquid Robotics and Ultra Electronic Undersea Sensors System Inc. (USSI), is the primary Sentinel payload, as it provides the greatest detection ranges. This sensor employs four omni-directional hydrophones in crossed-dipole array with a single centred reference element, to detect and estimate a line of bearing to surface contacts, Fig 4 (left). The hydrophones are sensitive to frequencies ranging from 100Hz to 4900Hz. The passive acoustic sensor is installed in a low-drag towbody, Fig. 4 (right), and towed behind the Wave Glider at a depth of 7 to 10 m. The tow-cable incorporates a motion isolation mechanism to minimize flow noise. The sensor employs broadband (BB) acoustic processing algorithms to make independent detections in eight linearly spaced 600Hz bands. Our BB processing provides detected source and ambient noise level estimates in each 600Hz band, as well as bearing estimates to the single loudest contact in each band. Bearing accuracy varies with detection band and signal to noise ratio. Narrowband processing algorithms are currently being developed to increase vessel detection ranges and improve classification acuity, to allow simultaneous detection of multiple targets via bearing discrimination, and to reject directional noise from shore facilities, off-shore platforms, and vehicle self-noise.

For the past several months, Liquid Robotics and Ultra USSI have been conducting at-sea engineering and functional testing of the broadband version of the Sentinel acoustic sensor in the vicinity of Kawaihae Harbor on the big island of Hawaii. Fig. 5 (left) shows
a typical surface vessel detection by a pair of Sentinel Wave Gliders. Overlapping lines of bearing allow the vessel’s position to be estimated. Bearing cones are shown with distinct colours and fixed widths for each 600 Hz detection band; lower frequency bands are shown with wider cones. Currently, the width of each band’s bearing cone is not (yet) adjusted to indicate instantaneous bearing uncertainty. Bearing-derived position estimates are then correlated with AIS ship positions (if any) received by the Wave Glider. Passive acoustic detections with Wave Glider Sentinel systems have been routinely demonstrated at ranges of 5 to 10 km for small vessels, and more than 25 km for loud or large ships (cruise ships, cargo barge tugboats, etc.). AIS reception ranges of 10 to 50 km are typical, depending upon a ship’s antenna height and atmospheric conditions (surface ducting).

Vessels are presumed to be operating legitimately if they are transmitting AIS messages. Nevertheless, if a boat passes within camera range of a Sentinel vehicle, a photo can be transmitted to shore for comparison with a free online database of vessel pictures to verify its self-reported identification. If a ship is found to be spoofing its AIS identification, then there is a high probability that it is engaged in some form of illegal activity.

Many smaller vessels are not equipped with AIS transmitters, and the absence of AIS messages from an acoustically detected surface contact is a common occurrence. In such cases, we can still track the ship via crossed lines of bearing, Fig. 5 (right), with a human operator monitoring the boat’s track and speed to determine if further scrutiny is warranted. Even when only a single Sentinel-equipped Wave Glider is present, changing SNR and lines of bearing and bearing rate of change can be combined to infer that a surface contact has likely crossed into a protection zone or that it is exhibiting other suspicious behaviours (e.g., making slow headway or loitering in an area that could be indicative of illegal fishing activity, or an unknown vessel making a fast transit in the direction of an offshore platform).

**MULTI-VEHICLE COORDINATED AUTONOMY**

To scale the PMMS to the degree necessary for persistent monitoring over large swathes of the ocean, human operators need to be able to define mission goals and monitor execution at the level of groups of vehicles. Towards this end, Liquid Robotics has developed a modular autonomy framework and robust control algorithms to autonomously
coordinate the actions of large numbers of Wave Glider vehicles to accomplish mission objectives with minimal interaction or oversight by human operators. The Multi-Vehicle Coordinated Autonomy (MVCA) program, a US Navy sponsored SBIR project, is based on an open architecture, modular autonomy framework that takes advantage of the “synoptic” centralized information flows to and from the Wave Glider’s shore-based command and control servers to facilitate efficient, coordinated control of multiple vehicles. The core autonomy framework is implemented in Java, an inherently portable and hardware-agnostic high-level computer language. This design choice allows the group-level autonomy software that runs on the shore command and control servers to also reside on the deployed USVs. Running the framework simultaneously on each vehicle and at the C2 server results in a much more flexible and powerful overall autonomy architecture, increasing the responsiveness of vehicle-level autonomous behaviors without sacrificing determinism in the centralized autonomous behaviors. The framework allows multiple autonomous behaviors to execute simultaneously without the need for a behavior hierarchy or other explicit mechanism to arbitrate conflicting commands between the behaviors.

Through the MVCA effort, we have developed autonomous behaviors for static obstacle avoidance, to navigate in the presence of exclusion zones of arbitrary shape and extent, to avoid dynamic obstacles (marine traffic), and to coordinate any number of vehicles in the autonomous formation and maintenance of fixed or translating lines and geometric patterns of vehicles. The ability to instantaneously re-test autonomy algorithm in simulation has been quite successful in exploring the effects of vehicle starting locations, the number of vehicles employed, and dynamically altering vehicle position in the face of changing conditions such as ocean current speed and direction, or the presence another vessel traversing the operating area. These new capabilities have been tested in sea trials with small arrays of three to five Wave Gliders, and the MVCA autonomy framework and the autonomous behaviors that we have built on top of it have proven to be quite flexible and robust.

CONCLUSIONS

The four components presented above – long-duration energy-harvesting unmanned maritime platforms, multi-modal communications infrastructure linking seafloor to surface to shore, a suite of sensors to support wide-area marine monitoring, and group-level autonomy for mission execution with large numbers vehicles – enable an affordable, modular approach for continuous ocean monitoring that is scalable in both the area covered and the overall cost expended to address missions with differing underlying economic impacts. The first two components of the Persistent Maritime Monitoring System have been proven through many vehicle-years of at-sea operations under all environmental conditions, including surviving hurricanes and typhoons. The third and fourth components are newly developed capabilities that have been tested in the last several months. We continue to develop our suite of Sentinel payload sensors and our multi-vehicle autonomous behaviours in preparation for an initial product release of these capabilities to Liquid Robotics customers in the third quarter of 2014. Future development will expand these offerings to include seafloor sensors tailored to the Sentinel mission set, and collaboration with shore- or platform-based small UAVs to enhance the PMMS’s vessel identification capabilities.
REFERENCES


VIRTUAL OCEAN TESTBED FOR AUTONOMOUS UNDERSEA SENSING NETWORKS

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Abstract: The MOOS-IvP Nested Autonomy architecture provides an open-source, behavior-based artificial intelligence framework for robust operation of undersea sensing networks with their inherently severe communication constraints, allowing for mission objectives to be achieved without the need for constant operator supervision and intervention. However, the reliance on a high level of autonomy requires extensive testing for performance demonstration and risk reduction ahead of extended operational deployments. To lower the cost and time required, a comprehensive simulation environment, the Virtual Ocean Testbed, has been developed, where the MOOS-IvP payload autonomy system on an arbitrary number of nodes is operated unchanged with the physical platform and the surrounding environment being replaced by physics-based simulators, including state-of-the-art ocean circulation models, high-fidelity acoustic propagation and scattering models, and advanced models for platform and sensor array dynamics. This paper describes the architecture of this Virtual Ocean Testbed and provides examples of its use for the planning of an experiment demonstrating the autonomous detection, classification and tracking of seabed objects using bistatic active acoustics.

Keywords: Autonomy, Network, Virtual Environments, Simulation, Bistatics, Detection, Tracking
1. INTRODUCTION

Being dependent on acoustic communication with a channel capacity many orders of magnitude smaller than the air and land-based equivalents, the operation of distributed undersea sensing and observation systems require a much higher level of autonomous, distributed data processing and control than land- and air-based equivalents. Nested Autonomy is a new command and control paradigm, inherently suited for the layered communication infrastructure provided by the low-bandwidth underwater acoustic communication and the intermittent RF connectivity [1]. Implemented using the open-source MOOS-IvP behavior-based, autonomous command and control architecture [2], it provides the fully integrated sensing, modeling and control that allows each platform to autonomously detect, classify, localize and track an episodic event in the ocean, without depending on any operator command and control. The prosecution of an event, such as the detection and tracking of a sub-sea volcanic plume or an oceanographic feature, or the acoustic detection, localization and classification of sea-bed objects, may be initiated by the operators through simple acoustic commands, or entirely autonomously by an onboard detection capability. The event information collected by each node in the network is reported back to the operators in a compact event report through the acoustic communication infrastructure. Collaborative processing and control is exploited when the communication channel allows, e.g. for collaborative tracking of a coastal front, or the tracking of marine mammals, but the fundamental principle of the nested autonomy paradigm is that each node must be capable of completing its mission objectives without requiring continued communication with the operators or other nodes.

The reliable and robust performance of this autonomy-centric operational paradigm introduces extremely strict requirements to the onboard autonomy system, in turn requiring extensive testing of individual software components and the autonomy system as a whole, preferably at sea. However, the available at-sea operation time is never sufficient for carrying out the thorough regression and unit testing that is required to ensure robust performance of the sensing network during extended field deployments, and the critical role of simulations in that regard is well established. The MOOS-IvP implementation of the autonomy system on the network nodes is based on the Payload Autonomy architecture [2], integrating the MOOS-IvP autonomy system with the sensor payload autonomy (the “backseat driver”) being separate from the native platform control (the “frontseat driver”), with a standard NMEA interface passing desired speed, heading and depth to the low-level vehicle control, and navigation information passed back to the payload autonomy system for fusion with the payload sensor information for use by the IvP-Helm decision-making. In addition, the MOOS-IvP autonomy architecture has a standard communication interface, based on the Goby autonomous communication software project [3,4]. These standard interfaces makes it possible to design the MOOS-IvP platform autonomy system without regard to whether it is operating in a real vehicle in the field or in a computer connected to simulators for the sensors, the communication infrastructure and the native platform dynamic control.

In support of several programs developing distributed ocean sensing concepts, MIT Laboratory for Autonomous Marine Sensing Systems (LAMSS) has established a generic Virtual Ocean Testbed, that allows for an arbitrary number of platform autonomy nodes to be operated in virtual experiments with a common ocean environment, e.g. generated by the MSEAS ocean circulation model [5], and which is then used by varying fidelity numerical models of the acoustic communication infrastructure, sonar sensor models, and the basic platform dynamics. By allowing extensive testing of the unchanged platform autonomy
software in this virtual environment, significant risk reduction is achieved at orders of magnitude less cost that would be required for pure at-sea testing.

2. PAYLOAD AUTONOMY ARCHITECTURE

The layering of the autonomy is continued on the individual platforms, where the intelligent autonomy allowing adaptive and collaborative sensing is handled in the payload, with the IvP-Helm issuing low-level commands for desired speed, heading and depth to the main vehicle computer. The MOOS-IvP Payload Autonomy architecture [1] provides an extensive legacy software base, which allows the research effort to immediately focus on the integration of the sensing, processing and modeling associated with the research objective at hand. In addition, a generic mission configuration infrastructure has been developed [2], with only a few mission-specific configuration parameters being changed between field deployments.

Figure 1 illustrates the modular structure of the MOOS-IvP Payload Autonomy architecture. The most critical component is the IvP-Helm – the “Captain of the Ship” - which is configured in a hierarchical mode structure, with each mode being associated with a particular mission component, such as “Search” or “Track”, and characterized by a set of pre-defined behaviors producing utility functions for speed, heading and depth. The autonomy mode can either be set through a command received from the operators through the communication infrastructure, or from the Mission Manager, which is a set of MOOS processes that manages the mission and the formation control, for example, maintaining a list of nodes in a local ‘cluster’ with whom the vehicle is in communication contact. The IvP-Helm applies its multi-objective optimization algorithm to the utility functions generated by the active behaviors to come up with a set of desired speed, heading and depth commands which are passed on to the lower level platform control. Another critical subsystem is the Sensor Control, which manages the sensor payload, i.e. for recording and processing acoustic data.
All the subsystems communicate using the MOOS publish-subscribe middleware, except for the communication with the “outside world”, which has three components: The AUV native control system (the frontseat), the payload sensors, and the acoustic communication. This architecture has the advantage that the entire autonomy system represented by the colored boxes in Figure 1 can be operated in the field with the actual hardware for these three outside connections, or with simulated versions of them, without any changes being necessary to the autonomy system itself. This feature is the key to the unique role of the complete system simulator described next, for system development and risk reduction. This architecture also enables the testing of any combination of physical and simulated external subsystems.

3. NETWORK COMMUNICATION, COMMAND AND CONTROL

The MOOS-IvP Payload Autonomy provides the fully integrated sensing, modeling and control that allows each platform, on its own or in collaboration with partners of opportunity, to autonomously detect, classify, localize and track (DCLT) an episodic, natural or human-created event, and subsequently report back to the operators. In spite of the high level of autonomy, a robust undersea communication infrastructure is critical to the operation of such networks. In contrast to air and land-based equivalents, the extremely limited bandwidth, latency and intermittency of underwater acoustic communication imposes severe requirements to the selectivity of message handling. Thus, contact and track reports for high-priority event, such as a detected chemical plume from a deep ocean vent, which may indicate an imminent volcanic eruption, must be transmitted to the system operators without delay. On the other hand, reports concerning less important events and platform status reports may be delayed without significant effects. Previous message handling systems for underwater communications have only a rigid, hard-coded queuing infrastructure, and do not support such advanced priority-based selectivity, hampering the type and amount of information that can be passed between cooperating nodes in the network.

Under the Goby GPL software project, a highly flexible and reliable communication infrastructure has been developed in support of autonomous sensing programs [2,3]. Goby has enabled the operation of a communication infrastructure with robust message handling for collaborative, autonomous sensing, as demonstrated in a handful of major recent field experiments. The Dynamic Compact Control Language (DCCL) used by the Goby communication stack allows for adequate navigation information to be packed with all other required message content, supporting complex collaborative maneuvers. The Goby/DCCL communication infrastructure forms the backbone of the MOOS-IvP Nested Autonomy Communication, Command and Control paradigm, shown schematically in Fig. 2.

Being based on established libraries of message handling software, the open source architecture of this Goby-MOOS communication stack lends itself directly to a wide range of military and civilian applications. It supports an arbitrary message suite and content without requirement of modifying software. All message encoding and decoding information is specified in a mission-unique configuration file written in the standard Google Protobuf format. Another feature of the Goby-MOOS infrastructure is a user configurable MAC scheme, including the option of automatically updating the communication sequencing when neighbor nodes appear and disappear from the local cluster.
The Goby communication infrastructure enables fully autonomously collaboration without any intervention by the operators, with each vehicle adapting its speed based on its current position and the position of the other vehicle extrapolated from the latest status reports. Such collaborative maneuvers would not be possible using traditional communication schemes, where navigation packets must be rigidly interleaved with messages containing data and command and control sequences. In contrast, the Dynamic Compact Control Language (DCCL) used by the Goby communication stack allows for adequate navigation information to be packed with all other required message content.

4. VIRTUAL OCEAN TESTBED

The MOOS-IvP Virtual Ocean testbed allows for 'operating' an arbitrary number of virtual sensing nodes in a common virtual 'ocean'. A comprehensive suite of high-fidelity sensor simulators allows each virtual vehicle to operate an autonomy system configuration which is identical to the one used on a field deployed vehicle. Further, the simulator incorporates a modem network simulator, providing realistic simulation of the underwater acoustic communication environment, including range-dependent transmission errors and collisions. Another unique component of the simulator is a suite of high-fidelity environmental acoustic simulators, allowing for generation of sensor level time series, incorporating environmental acoustic effects such as boundary interactions, refraction, noise and reverberation. The acoustic sensor models are closely tied to the platform and sensor dynamic simulators, including a model of the coupled dynamics of an AUV and a towed acoustic array.

Figure 3 shows a "Virtual Ocean" equivalent of the multi-node Undersea Network in Fig. 2. The real ocean and the AUV control system, sensors, and modems are replaced by simulated versions, all using the MOOS uField virtual network control infrastructure. A
shore-based MOOS environment controls the network communication, and has tools for replacing the physical sensors with a simulated measurement, and for managing the communication to be consistent with the constraints observed in the field, both in terms of bandwidth and range.

For example, as illustrated in Fig. 3, to provide realistic platform dynamics, e.g. in response to spatial and temporal variability of the currents, the uField Network Control keeps track of where all the nodes are currently located through a continuous stream of NODE_REPORTs from the virtual vehicle’s “frontseat”, and then requests from the ocean circulation model a current vector for that location, and transmits it to the AUV MOOS community, which is running a dynamic model for the vehicle instead of the physical “frontseat”. This whole process happens continuously, completely transparent to the MOOS-IvP Autonomy system.

![Diagram](image)

**Fig. 3.** Virtual Ocean with high-fidelity simulation of platform dynamics, acoustic communication and the physics of the sensor system. The uField MOOS community manages the simulation consistently for all nodes, including collaborative sensing such as bistatics.

Similarly, Fig. 3 illustrates the simulation of the acoustic networking in the Virtual Ocean. The AUV Mako in this case wants to broadcast a message to any neighbor nodes capable of receiving. The message is sent to Goby-2, who has a dedicated driver for the physical layer replaced by uField, packages the message into an uField NODE_MESSAGE. The message is sent to the uField community, where a communication model, simulating latency, collisions, range, etc., decides which other nodes will receive the message, and it sends it to those vehicle communities, where Goby-2 handles it as it would handle the same message in the real ocean. The communication model can be configured for different levels of fidelity. In the simplest mode, it is simply blocking messages destined for nodes outside a configured maximum range, or it may request a transmission loss from a model such as Bellhop, and blocking messages to nodes with transmission loss above a configured threshold. For the highest level of fidelity, the message is sent to a physical modem emulator. The analog signal is then digitized and ‘transmitted’ to a receiving modem emulator via an
environmental acoustic model such as Bellhop. The possibility of operating the autonomy system under such highly realistic communication conditions is critical to ensuring the robustness of a distributed sensing system for extended deployments. Note that in this context, the topside command and control is handled as a network node, similarly to all other sensor nodes, allowing also the topside MOOS community to be operated unchanged from its field equivalent.

Finally, Fig. 3 illustrates how an active sonar system on one of the sensing nodes is handled in the virtual Ocean. A standard ping command in the format defined by the interface control document for the actual sonar is sent to the sonar simulation module, which then in turn translates this into an ACTIVE SONAR REQUEST which is sent to the acoustic modeling module uSimActiveSonar in the uField community with sonar parameters such as source/receiver locations, pulse type, frequency and bandwidth. The simulation model then requests the computation of the acoustic propagation to and from simulated targets and the ocean boundaries from the generic iBellhop interface to the legacy Bellhop propagation model. Once the computation is completed, the sonar simulator combines the computed transfer functions with the appropriate scattering and reverberation models and generates the element-level timeseries of the received signals. Once completed, it sends an ACTIVE SONAR RESPONSE back to the node autonomy system with the filename containing the signals, which are then available for the onboard sonar processing modules for target detection, classification, localization and tracking (DCLT).

Fig. 4. Virtual experiment demonstrating autonomous detection and tracking of seabed targets using bistatic active acoustics. Unicorn’s current location is the lower left corner of the hexagonal survey.

An example of the application of the high-fidelity Virtual Ocean Testbed, Fig. 4 shows the results of a virtual experiment carried out as part of the planning for the Bayex’14 experiment in Panama City, FL in June 2014, where the AUV Unicorn, equipped with a 16-element ‘swordfish’ array will perform an extensive sampling of the bistatically scattered field from a spherical and a cylindrical target on the seabed, insonified by a broadband acoustic source on a fixed tower. In this virtual experiment, Unicorn is performing a hexagonal survey surrounding the two targets, located 70-80 m from the source, among an artificial reverberant field, as shown in the operational map to the right. The left frame shows the real-time display of the active sonar simulator and array processing, with the contour plot
showing the beamformed array response with both the direct signal, the reverberation, and the late scattering from the two targets clearly visible. The map also shows the two target tracks developed by the ‘onboard’ processing.

5. CONCLUSIONS

The MOOS-IvP Nested Autonomy architecture lends itself naturally to incorporation into a high-fidelity simulation environment, where physical sensors, communication modems and vehicles are replaced by their virtual equivalents, of varying level of fidelity. Such a Virtual Ocean Testbed has been developed and used extensive in parallel to and in preparation of at-sea experiments and demonstrations, and has been instrumental in several recent successful demonstrations of fully autonomous, collaborative and adaptive oceanographic and acoustic sensing with networks of autonomous underwater vehicles.

6. ACKNOWLEDGEMENTS

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REFERENCES

Session 32

Vector Sensors: Development and Applications

Organizers: Tuncay Akal, Sergio Jesus and Jean-Pierre Hermand
AN EXPERIMENTAL STUDY ON DEMON SPECTRUM
DIRECTION ESTIMATION OF MULTI-TARGET WITH A LOW
FREQUENCY VECTOR HYDROPHONE

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Abstract: Due to the periodic rotation of the ship propellers, its radiated noise will carry rich discrete components including shaft and blade frequency and their multiplications information. And these discrete components may be modulated by the continuous spectrum involved in the radiated noise and produce the DEMON spectrum. The DEMON spectrum is actually a kind of modulated spectrum of the ship radiated noise, from which the information about the shaft and blade can be extracted. What’s more, the information can be used to distinguish the targets of interest. Owing to the DOA estimation ability of a single vector sensor, such discrete component can be used to distinguish the targets distributed in different directions. Most of the existing DEMON methods make use of the high frequency (above 1 kHz) information, from which the blade information is obtained. In this paper, the study focuses on the low frequency DEMON spectrum extraction for multi-target and their DOA estimation with a single vector sensor. The sea trial data acquired by a submerged buoy was processed using the above method. Results indicated that DEMON spectrum is rich in low frequency, from which the shaft component and DOA could be determined. Compared with the method based on cross-spectrum DOA estimation using a vector sensor, the presented method could achieve better ability for multi-target discriminate and DOA estimation.

Keywords: Low frequency, vector hydrophone, direction estimation, DEMON
1. INTRODUCTION

Multi-target separation is one of the hottest topics in underwater acoustics. The solution has been described in the literature[1-4]. The difference between the targets is the absolute rule to separate one from another. Generally, three methods could be used to achieve this, which were called time domain method, frequency domain method and spatial domain method. In addition, DEMON spectrum of the ship noise, representing rich propeller and blade information, can be used to separate ship targets from each other as a frequency domain method. A. Sedunov [1] and his colleagues developed a cross-correlation method which had a higher time resolution to separate one ship target from another in the time domain successfully. Because of the lack of the spectrum features information of the targets, the frequency domain method was almost never used alone except in navy sonar system. Most of the time, frequency domain method was used together with the spatial domain method. If different frequency components correspond to different azimuths, the targets can be distinguished easily.

The purpose of this paper is to analyse the method of DEMON spectrum direction estimation with a single vector hydrophone, which could be used to separate multi-target, and to show some experiment results. The remaindering parts of this paper are organized as follows: the method of DEMON spectrum direction estimation, and some related details are given in the section II; then, the setup of the sea experiment and some results are given and discussed in section III; finally conclusions are drawn.

2. DEMON SPECTRUM DIRECTION ESTIMATION WITH A SINGLE VECTOR HYDROPHONE

2.1 PRINCIPLE OF DEMON SPECTRUM DIRECTION ESTIMATION

Traditionally, the acoustic signal is received by one hydrophone or an array consists of many hydrophones which could only represent the pressure. With the advent of vector hydrophone, the higher order components (usually referred to velocity) of the acoustic field of the same spatial point could be measured additionally at the same time. The direction that was estimated by an array now can be derived by a single vector hydrophone because velocity is a vector quantity[7-8]. What’s more, the vector hydrophone has great advantages in low frequency since its directivity is independent of frequency.

The ship noise is modulated by the rotation of propeller, so the signal radiated by a ship and received by a vector hydrophone can be expressed as follows:

\[ p(t) = \left(1 + \sum_{n=1}^{N} m_n \cos(n\Omega t) \right) s_0(t) \]  

\[ v_x(t) = \left(1 + \sum_{n=1}^{N} m_n \cos(n\Omega t) \right) s_0(t) \cos \alpha \]  

\[ v_y(t) = \left(1 + \sum_{n=1}^{N} m_n \cos(n\Omega t) \right) s_0(t) \sin \alpha \]  

Where \( p(t) \) is the time series of pressure, \( v_x(t) \) and \( v_y(t) \) are the time series of velocity corresponding to \( x \)-axis and \( y \)-axis respectively, \( s_0(t) \) is the time series of pre-modulated ship noise, \( \Omega \) is the radical frequency, \( m_n \) is the modulating coefficient of \( n \)th harmonic, \( N \) is the number of harmonics, \( \alpha \) is the direction of the ship target.
Without generality, we let \( N = 1 \), and \( s_0(t) = \sin(\omega_0 t) \), we may derive
\[
p(t) = (1 + m_1 \cos(\Omega t))\sin(\omega_0 t) \tag{4}
\]
\[
v_x(t) = (1 + m_1 \cos(\Omega t))\sin(\omega_0 t) \cos \alpha \tag{5}
\]
\[
v_y(t) = (1 + m_1 \cos(\Omega t))\sin(\omega_0 t) \sin \alpha \tag{6}
\]
Take the absolute of the signal and pass a low pass filter we get
\[
|p(t)|_{LP} = 1 + m_1 \cos(\Omega t) \tag{7}
\]
\[
|v_x(t)|_{LP} = (1 + m_1 \cos(\Omega t))|\cos \alpha| \tag{8}
\]
\[
|v_y(t)|_{LP} = (1 + m_1 \cos(\Omega t))|\sin \alpha| \tag{9}
\]
Then compute FFT of the time series, we get
\[
P(f) = \text{FFT} \{p(t)|_{LP}\} \tag{10}
\]
\[
V_x(f) = \text{FFT} \{v_x(t)|_{LP}\} \tag{11}
\]
\[
V_y(f) = \text{FFT} \{v_y(t)|_{LP}\} \tag{12}
\]
The acoustic intensity could be derived by multiplying the pressure spectrum and conjugate of the velocity and then taking the real part:
\[
I_x(f) = \text{Re}\{P(f)*V_x(f)^*\} \tag{13}
\]
\[
I_y(f) = \text{Re}\{P(f)*V_y(f)^*\} \tag{14}
\]
Thus the relationship between the direction of ship target and acoustic intensity could be expressed as:
\[
|\tan \alpha| = \frac{I_y(f)}{I_x(f)} \tag{15}
\]
Obviously, the direction solution, which is actually only one direction solution, from the equation is ambiguous with four possible quadrants. The method to remove the ambiguity will be given in the next section.

2.2 REMOVAL OF DIRECTION AMBIGUITY

To determine the quadrant of the ship target, the heart directivity of the vector hydrophone could be utilized. According to the mathematical model of the received signal, the combined signal \( v_x(t) + p(t) \) has the directivity

\[
R_{v1}(\alpha) = \cos \alpha + 1 \tag{16}
\]
and \( v_x(t) - p(t) \) has the directivity

\[
R_{v2}(\alpha) = \cos \alpha - 1 \tag{17}
\]
The directivity \( R_{v1}(\alpha) \) and \( R_{v2}(\alpha) \), are called cardiac curve. The directions of two curves are opposite. While the maximum of one cardiac curve points to the target, the null of another cardiac curve points to the target. In other words, we derive more energy with one directivity curve than another.

Let
\[
R_v(\alpha) = R_{v1}(\alpha) - R_{v2}(\alpha) \tag{18}
\]
Then if \( R_v(\alpha) > 0 \), the target is in quadrant I or quadrant IV.
If \( R_v(\alpha) < 0 \), the target is in quadrant II or quadrant III.
Similarly, we could define
\[
R_{y1}(\alpha) = \sin \alpha + 1 \tag{19}
\]
\[
R_{y2}(\alpha) = \sin \alpha - 1 \tag{20}
\]
If \( R_y(\alpha) > 0 \), the target is in quadrant I or quadrant II.

If \( R_y(\alpha) < 0 \), the target is in quadrant III or quadrant IV.

So the quadrant could be determined by the sign of \( R_x(\alpha) \) and \( R_y(\alpha) \).

2.3 WHY LOW FREQUENCY?

To extract the DEMON spectrum from ship noise, besides the serious choice of the extraction method, another important thing that should be paid attention to is the choice of analyzed frequency band. Generally, people preferred to analyze the DEMON spectrum from the signal whose frequency band is between several hundred Hz to several kHz. However, in this paper, we analyze the signal of several tens of Hz to several hundreds of Hz. Why? Generally, there are two reasons:

Firstly, the main energy of the ship noise concentrates in several tens of Hz to several hundreds of Hz. Fig.1 showed one of the Sea trial results of pressure spectrum at different time. The intensity of the spectrum was indicated by the value of colour. When the ship passed by the vector hydrophone, the intensity of low frequency became much more prominent. This fact has also been demonstrated by many authors in their literature (see also e.g. [5-6], etc).

Secondly, the vector hydrophone has great advantage over low frequency[8-9]. The operating frequency of array of multiple pressure elements is usually above several hundred Hz because the spatial aperture is limited and the requirements of more independent elements for higher direction estimation precision. Yet vector hydrophone could estimate the direction of a target at very low frequency with low cost of the increment of the aperture since its directivity is independent of frequency.

3. EXPERIMENTS IN THE YELLOW SEA OF CHINA

3.1 SETUP OF THE EXPERIMENTS

The experiment was conducted by a scientific survey ship in July of 2010, at Yellow sea of China. The depth of the sea site is 40 meters and the traffic is heavy. The sound velocity profile was shown in the right side of Fig.2. After deploying the vector hydrophone to the bottom of the sea, the survey ship moved 10km away and monitoring the traffic information of the surface ship with AIS and GPS system. The electronic unit connected with the vector hydrophone recording and storing the noise data.
3.2 EXPERIMENTS RESULTS

Two typical results of DEMON spectrum direction estimation are given in this section: one is with one ship target only and another is with four ship target.

Fig.3 and Fig.4 gave the one ship target’s spectrogram and direction estimation of DEMON respectively. The DEMON spectrum lines are stable and their signal to noise ratio is high enough to estimate their direction. The direction estimation results were in accordance with the monitoring results of AIS.

Fig.5 and Fig.6 gave the four ship targets’ spectrogram and direction estimation of DEMON respectively. Though the direction of multi-target could be estimated as shown in the figure, there is some interference between targets. At the time of 7700s, we could see that the DEMON spectrum of one ship disappeared suddenly and DEMON spectrum...
of another ship appeared, this might because the percentage of the ship noise to the total noise changed.

3.3 DISCUSSION OF THE INTERFERENCE OF MULTI-TARGET

When multi-target appeared simultaneously with the same modulating coefficients and same sound level, it’s difficult to separate them with the absolute DEMON spectrum extraction method merely because of the interference between them. However, if combined with time separation (e.g. correlation method) or direction estimation, the separation of multi-target is possible.

When the noise level of one ship target is much higher than the others, only the DEMON spectrum of the higher one could be extracted because the DEMON extracting method requires relative high signal to noise ratio.

4. CONCLUSION

The DEMON spectrum showed the character of the shaft or blade frequency of the ship noise and could be a useful quantity to separate one ship target from another. Combined with the ability of direction estimation with a single vector hydrophone, multi-target could be separated and it was verified by the sea experiment results.

5. ACKNOWLEDGEMENTS

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REFERENCES

SIGNAL PROCESSING FOR CIRCULAR VECTOR-SENSOR ARRAY MOUNTED AROUND A CYLINDRICAL BAFFLE

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Abstract: A method of the Acoustic Vector-Sensor Array signal Processing for a uniform circular acoustic Vector-Sensor Array (UCAVSA) mounted around a cylindrical baffle is presented. Using the elastic thin shell theory, the analytic expressions for the scattered pressure and particle velocity are derived. It is found that the pressure and the particle velocity fields near the surface of the cylindrical baffle are characterized by complex interference structure. Then the total pressure field and the total particle velocity field near the surface of the cylindrical baffle are analyzed theoretically by applying the method of spatial Fourier transform. The so-called modal vector-sensor array signal processing algorithm, which is based on the decomposed wavefield representations, for the UCAVSA mounted around the cylindrical baffle is proposed. Simulation and experimental results show that the UCAVSA mounted around the cylindrical baffle has distinct advantages over the same manifold of traditional uniform circular pressure-sensor array (UCPSA). It is pointed out that the acoustic Vector-Sensor (AVS) could be used under the condition of the cylindrical baffle and that the UCAVSA mounted around the cylindrical baffle could also combine the anti-noise performance of the AVS with spatial resolution performance of array system by means of modal vector-sensor array signal processing algorithms.

Keywords: acoustic vector sensor, cylindrical baffle, modal vector-sensor array signal processing algorithm
1. INTRODUCTION

The acoustic vector sensor (AVS) has been successfully applied to marine sonar buoys, towed array sonar and other acoustic devices. These works have only considered the acoustic vector sensors in free space; however, when the acoustic vector sensors are mounted on the acoustic baffle, due to the scattering of the acoustic baffle, the distortions of the acoustic fields will render the original vector signal processing method based on the free space no longer applicable[1,2]. Recently, some scholars have been involved in the solution of this problem. Hawkes et al. considered the passive direction direction-of-arrival estimation problem using arrays of acoustic vector sensors near a reflecting boundary [3]. Nan Zou introduced the acoustic signal model of a circular vector-sensor array which senses both the radial and tangential components of the acoustic field gradient, but did not include sound pressure component [4]. So far, there have been no studies on acoustic vector characteristics of near fields scattered by a cylindrical baffle and the signal processing method for circular Vector-Sensor Array mounted around a cylindrical baffle.

This paper takes the cylindrical baffle, which is the typical sonar baffle on ships, as a model. Firstly, the total pressure field and the total particle velocity field near the surface of the cylindrical baffle are analyzed theoretically. Then the coherent signal processing algorithm for the pressure signal and the particle velocity signal is proposed. Simulation and experimental results show that the UCAVSA mounted around the cylindrical baffle has distinct advantages over the same manifold of traditional UCPSA.

2. MEASUREMENT MODEL

We consider a UCAVSA mounted around a cylindrical baffle. The cylindrical baffle is submerged in an acoustic medium of density $\rho$ and velocity $c$; the interior of the baffle is another acoustic medium of density $\rho_0$ and velocity $c_0$. Let $a$ and $l$, respectively, represent the radius and the half-length of the cylinder baffle. The array elements of the UCAVSA are uniformly distributed over the circumference in the $xOy$ plane, with the distance from the surface of the
cylindrical baffle \( d \). A unit plane wave impinging from \((\theta_0, \phi_0)\) is emitted by a source “S” which is located in the far-field of the UCAVSA. Source elevation angle \( \theta_0 \in [0, \pi/2] \) is measured down from the \( z \) axis, and azimuth angle \( \phi_0 \in [0, 2\pi] \) is measured counterclockwise from the \( x \) axis, as shown in Figure 1.

The total acoustic wavefields at the UCAVSA can be expressed as [5]:

\[
p(r, \varphi, z) = \exp(\imath k z) \sum_{n=-\infty}^{\infty} b_n \exp[\imath n (\varphi - \varphi_0)]
\]

\[
vr(r, \varphi, z) = \exp(\imath k z) \sum_{n=-\infty}^{\infty} (-j) b_n \exp[\imath n (\varphi - \varphi_0)] / k \rho c
\]

\[
v\varphi(r, \varphi, z) = \exp(\imath k z) \sum_{n=-\infty}^{\infty} n b_n \exp[\imath n (\varphi - \varphi_0)] / r k \rho c
\]

where

\[
b_n = j^n \{ J_n(kr) - \frac{J_n(kr) - H_n^{(1)}(kr)}{H_n^{(1)}(kr)} \} + \sum_{p=1}^{\infty} \frac{2 \rho \omega}{j \pi \alpha_p k a H_n^{(1)}(\alpha_p a) H_n^{(1)}(k a)} F(k, l, k, l) \exp(-j \varphi_0)
\]

\[
\alpha_p = \left\{ j \left[ k^2 - k_p^2, k > k_p \right] \right\}, \quad b_n = \left\{ \frac{2 \rho \omega}{j \pi \alpha_p k a H_n^{(1)}(\alpha_p a) H_n^{(1)}(k a)} F(k, l, k, l) \right\} \exp(-j \varphi_0)
\]

\[
F(k, l, k, l) = \frac{\sin(k l + k l)}{k l + k l} \exp(j k l) - \frac{\sin(k l - k l)}{k l - k l} \exp(-j k l), \quad Z_n^{(1)} = \frac{j \omega a H_n^{(1)}(\alpha_p a)}{\alpha_p H_n^{(1)}(\alpha_p a)}
\]

\[
, \quad \alpha = k a, \quad Z_n^2 = -\frac{j \omega a \Omega}{\alpha_p J_n^{(1)}(\alpha_p a)}, \quad \Omega = k a, \quad \mu = \frac{1 - \sigma}{2}, \quad \mu' = \frac{1 + \sigma}{2}, \quad c_s = \sqrt{\frac{E}{\rho_s}(1 - \sigma^2)}
\]

\[ k_s = \frac{\omega}{c_s}, \quad \sigma \text{ is Poisson’s constant, } \rho_s \text{ is the shell density, } E \text{ is the Young’s modulus of the shell material, } k = \omega/c \text{ is the exterior acoustic wave number, } k_s = k \sin \theta_0, \quad k_z = k \cos \theta_0, \quad J_n(.) \text{ is the cylindrical Bessel function of order } n, \]

\[ H_n^{(1)}(.) \text{ is the cylindrical Hankel functions of order } n \text{ of the first kind}.\]

\[ b_n' \text{ is the derivative from } b_n \text{ with respect to } r.\]

3. ACOUSTIC FIELDS REPRESENTATION BY WAVEFIELD DECOMPOSITION

Wavefield decomposition is a technique that decomposes a wavefield into spatially orthogonal eigen-solutions to the acoustic wave equation in a coordinate system that best suits the geometry of the aperture under consideration [6, 7]. For simplicity, only normal wave incidence with respect to the \( z \) axis is considered in the following discussions, i.e. \( \theta_0 = \pi/2 \). The two-dimensional wavefield at the UCAVSA could be decomposed into an orthogonal set of eigen-funtions of the
acoustic wave equation in cylindrical coordinates. Let $P_n$ be the Fourier transform in $\phi$ of the acoustic pressure at $r$:

$$P_n = \frac{1}{2\pi} \int_0^{2\pi} p(r, \phi, z)e^{-jn\phi_0} d\phi = b_n \exp(-jn\phi_0)$$  \hspace{1cm} (2)

the symbol $P_n$ is referred to as an eigen-modal of order $n$.

\begin{figure}
\centering
\includegraphics[width=\textwidth]{modal.png}
\caption{Modal amplitude versus modal order.}
\end{figure}

Figure 2 shows the modal magnitude response over modal order $n$. Each curve corresponds to a different $kr$ for the cylindrical baffle ($a=0.25m$, $l=1m$, $d=0.09m$). We can see that the curves have obvious amplitude fluctuations caused by the elastic resonance, which differs from the rigid boundary condition. For $kr=6$ we can see that there are about 7 dominant modals. The higher $kr$ gets, the more modals are present. For each $kr$ there are roughly $[1.2 \times kr]$ modals present. Therefore the total acoustic pressure at the UCAVSA can be simplified:

$$p(r, \phi, z) = \sum_{n=-K}^{K} P_n \exp(jn\phi)$$  \hspace{1cm} (3)

where $K = [1.2 \times kr]$. Similar to the total acoustic pressure, the definitions of the eigen-modals of order $n$ for the radial and circumferential particle velocity at the UCAVSA are:

$$\begin{cases}
VR_n = (-j)\frac{h_n \exp(-jn\phi_0)}{\omega \rho} \\
V\Phi_n = nb_n \exp(-jn\phi_0)/r \omega \rho
\end{cases}$$  \hspace{1cm} (4)

The total radial and circumferential particle velocity at the UCAVSA can also be simplified:

$$\begin{cases}
vr(r, \phi, z) = \sum_{n=-K}^{K} VR_n \exp(jn\phi) \\
v\phi(r, \phi, z) = \sum_{n=-K}^{K} V\Phi_n \exp(jn\phi)
\end{cases}$$  \hspace{1cm} (5)
4. APPLICATIONS OF MODAL ARRAY SIGNAL PROCESSING FOR THE UCVSA

Suppose that plane waves from \( N \) distant acoustic sources are impinging on the UCAVSA composed of \( M \) elements. The signals received at the array elements are linear combinations of the \( N \) incident acoustic sources. Thus, the signals received by the UCAVSA at time \( t \), using eq.(4)-(5), could be expressed as:

\[
\begin{align*}
\mathbf{p}(t) &= \mathbf{F}_p \mathbf{A}(\phi) s(t) + \mathbf{n}_p(t), \\
\mathbf{v}_r(t) &= \mathbf{F}_{vr} \mathbf{A}(\phi) s(t) + \mathbf{n}_{vr}(t), \\
\mathbf{v}_\phi(t) &= \mathbf{F}_{v\phi} \mathbf{A}(\phi) s(t) + \mathbf{n}_{v\phi}(t).
\end{align*}
\]

(6)

where \( \mathbf{p}(t)=[p_1(t),\ldots,p_M(t)]^T \), \( \mathbf{v}_r(t)=[v_{r1}(t),\ldots,v_{rM}(t)]^T \) and \( \mathbf{v}_\phi(t)=[v_{\phi1}(t),\ldots,v_{\phiM}(t)]^T \) are the outputs of the pressure sensor and particle velocity sensors, \( \mathbf{n}_p(t)=[n_{p1}(t),\ldots,n_{pM}(t)]^T \) and \( \mathbf{n}_{vr}(t)=[n_{vr1}(t),\ldots,n_{vrM}(t)]^T \) and \( \mathbf{n}_{v\phi}(t)=[n_{v\phi1}(t),\ldots,n_{v\phiM}(t)]^T \) represent noise. The source vector is defined as \( \mathbf{s}(t)=[s_1(t),\ldots,s_N(t)]^T \). The noise field is assumed to be independent of the source signals. The other matrixes are defined as below:

\[
\mathbf{F}=[\mathbf{w}_K,\mathbf{w}_{K1},\ldots,\mathbf{w}_K], \mathbf{A}(\phi) = [\mathbf{a}(\phi_1),\ldots,\mathbf{a}(\phi_N)], \mathbf{a}(\phi_q) = [\exp(-jK\phi_q),\ldots,\exp(jK\phi_q)]^T, \\
\mathbf{B}_p = \text{diag}[b_{-K},\ldots,b_K], \mathbf{B}_{vr} = \text{diag}[b_{-K}/jK\rho_c,\ldots,b_K/jK\rho_c], \\
\mathbf{B}_{v\phi} = \text{diag}[-Kb_{-K}/rK\rho_c,\ldots,Kb_K/rK\rho_c], \mathbf{w}_q = [1,\exp(-j2\pi q/M),\ldots,\exp(-j2\pi q(M-1)/M)]^T.
\]

The spatial discrete Fourier transform along the aperture of the acoustic pressure at time-instance \( t \) can be written as

\[
\mathbf{u}(t) = \mathbf{F}^H \mathbf{p}(t) = \mathbf{M} \mathbf{B}_p \mathbf{A}(\phi) s(t) + \mathbf{M} \mathbf{B}_p \mathbf{F}^H \mathbf{n}_p(t)
\]

(7)

eq(7) can be modified as,

\[
\frac{1}{M} \mathbf{B}_p^{-1} \mathbf{F}^H \mathbf{p}(t) = \mathbf{A}(\phi) s(t) + \frac{1}{M} \mathbf{B}_p^{-1} \mathbf{F}^H \mathbf{n}_p(t)
\]

(8)

The transformation matrix \( \mathbf{T}_p \) is defined as,

\[
\mathbf{T}_p = \frac{1}{M} \mathbf{B}_p^{-1} \mathbf{F}^H
\]

(9)

eq(8) shows that the transformation matrix \( \mathbf{T}_p \) can transform the actual array vector \( \mathbf{p} \) from element space into modal space,

\[
\mathbf{p}_e(t) = \mathbf{T}_p \mathbf{p}(t) = \mathbf{A}(\phi) s(t) + \mathbf{T}_p \mathbf{n}_p(t)
\]

(10)

Similarly, the transformation matrix for the radial and circumferential particle velocity is defined by:
The signals for the radial and circumferential particle velocity in modal space can be yielded,

\[
\begin{align*}
T_{rr} &= \frac{1}{M} B_{rr}^{-1} F^{H} , \\
T_{r\phi} &= \frac{1}{M} B_{r\phi}^{-1} F^{H} .
\end{align*}
\]  

The signals for the radial and circumferential particle velocity in modal space can be yielded,

\[
\begin{align*}
v_{rr}(t) &= T_{rr} v_r(t) = A(\phi) s(t) + T_{r\phi} n_r(t) \\
v_{r\phi}(t) &= T_{r\phi} v_\phi(t) = A(\phi) s(t) + T_{r\phi} n_\phi(t)
\end{align*}
\]  

Form \((2K+1)\times(2K+1)\) modal space covariance matrix of the UCAVSA as,

\[
R_{rr\phi\phi} = E[p_r(t)v_r(t)^H + p_\phi(t)v_\phi(t)^H]
\]  

In particular, the Bartlett beamformer, the conventional beamforming direction estimator, is obtained

\[
P_{BF} = \frac{q^{H}(\phi) R_{rr\phi\phi} (\phi) a(\phi)^{H}}{u^{H}(\phi)a(\phi)}
\]  

The steering vector \(a(\phi)\) takes the form \(a(\phi) = [\exp(-jK\phi) \cdots \exp(jK\phi)]^T\).

5. SIMULATIONS AND EXPERIMENTAL RESULTS

The simulations are done with a UCAVSA of 8 elements mounted around a cylindrical baffle. The elements of the UCAVSA are uniformly distributed over the circumference in the \(xOy\) plane, with the distance from the surface of the cylindrical baffle \(d = 0.09\) m. The parameters of the cylindrical baffle correspond to the values used in the simulations and experiments: radius \(a = 0.25\) m, a total length \(2l = 1\) m, shell thickness \(h = 0.002\) m. The elastic parameters are that of steel with \(E = 2.1 \times 10^{11}\) Pa, \(\sigma = 0.3\), and \(\rho_s = 7850\) kg/m\(^3\) while the external fluid is water with \(c = 1500\) m/s and \(\rho = 1000\) kg/m\(^3\), the interior fluid is air with \(c_0 = 346\) m/s and \(\rho_0 = 1.29\) kg/m\(^3\). The incident unit harmonic plane wave originates from \(\theta_0 = 90^\circ\) and \(\phi_0 = 45^\circ\).

Figure 3 shows the simulated beampattern of the Bartlett beamformer for the UCAVSA and the beampattern of the Bartlett beamformer for the UCPSA at frequencies \(f = 3150\) Hz and \(f = 4000\) Hz. Figure 4 shows the outputs of Bartlett beamformers for the UCAVSA and the UCPSA with different SNR at frequent \(f = 4000\) Hz.
Figure 3  Simulated beampatterns. (a) $f = 3150\text{Hz}$. (b) $f = 4000\text{Hz}$.

Figure 4  Simulated output. (a) $\text{SNR} = -10\text{dB}$. (b) $\text{SNR} = -20\text{dB}$.

Figure 5 shows the measured beampatterns of the two arrays at frequencies $f = 3150\text{Hz}$ and $f = 4000\text{Hz}$. Figure 6 shows the target tracking results of the UCAVSA and UCPSA.

Figure 5  Measured beampatterns. (a) $f = 3150\text{Hz}$. (b) $f = 4000\text{Hz}$.

Figure 6  Target tracking results (a) UCPSA. (b) UCAVSA.
6. CONCLUSION

DOA estimation for a uniform circular acoustic Vector-Sensor Array (UCAVSA) mounted around a cylindrical baffle is investigated in this paper. The modal vector-sensor array signal processing algorithm, which is based on the Wavefield decomposition techniques, for the UCAVSA mounted around the cylindrical baffle is proposed. Compared to the same manifold of the UCPSA, the UCAVSA mounted around the cylindrical baffle has narrower beamwidth and better anti-noise performance. It is concluded that the AVS can be used under the condition of the cylindrical baffle and that the UCAVSA mounted around the cylindrical baffle can also combine the anti-noise performance of the AVS with spatial resolution performance of array system.

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BROADBAND DOA ESTIMATION IN PHASE MODAL
SPACE FOR CIRCULAR ACOUSTIC VECTOR-SENSOR
ARRAY

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Abstract: An approach to Broadband direction of arrival (DOA) estimation for a uniform circular acoustic Vector-Sensor Array (UCAVSA) is proposed. The pressure field and the particle velocity field are decomposed into an orthogonal set of phase modal s by applying the method of wavefield decomposition. The method to construct pretreatment matrix, which transform the received signals from element space to phase modal space, is proposed for broadband farfield signals. Simulation results show that the UCAVSA has distinct advantages over the same manifold of traditional uniform circular pressure-sensor array (UCPSA) in anti-noise performance, resolution and accuracy. The method is based on the principle of coherency between pressure and particle velocity, which can suppress interference in isotropic noise field. The algorithm could combine the anti-noise performance of the Vector-Sensor with spatial resolution performance of array system to solve the problem of high-resolution DOA estimation of remote targets for UCAVSA.

Keywords: Broadband, direction of arrival, phase modal
1. INTRODUCTION

The Acoustic vector sensor (AVS) technology is research focus in the last two decades. The AVS has been successfully applied to marine sonar buoys, towed array sonar and other acoustic devices [1-3]. But most studies limited to uniform linear array (ULA).

This paper takes uniform circular acoustic Vector-Sensor Array (UCAVSA) as a model. Firstly, the pressure field and the particle velocity field are decomposed into an orthogonal set of phase modals by applying the method of wavefield decomposition. Secondly, the method to construct pretreatment matrix, which transform the received signals from element space to phase modal space, is proposed for broadband farfield signals. Then the coherent signal processing algorithm for the pressure signal and the particle velocity signal is proposed. Simulation and experimental results show that the UCAVSA has distinct advantages over the same manifold of traditional uniform circular pressure-sensor array (UCPSA) in anti-noise performance, resolution and accuracy.

2. MEASUREMENT MODEL

Consider a UCAVSA in the xOy plane. The radius of UCAVSA is r. The array elements of the UCAVSA are uniformly distributed in the xOy plane. Suppose that N far field broadband sources are impinging on the UCAVSA composed of M elements. Source elevation angle $\theta_n \in [0, \pi/2]$ is measured down from the z axis, and azimuth angle $\varphi_n \in [0, 2\pi]$ is measured counterclockwise from the x axis, as shown in Figure 1. For simplicity, only normal wave incidence with respect to the z axis is considered in the following discussions.

Figure 1 Geometric model
Consider a UCAVSA in the xOy plane. The radius of UCAVSA is r. The array elements of the UCAVSA are uniformly distributed in the xOy plane. Suppose that N far field broadband sources are impinging on the UCAVSA composed of M elements. Source elevation angle $\theta_n \in [0, \pi/2]$ is measured down from the z axis, and azimuth angle $\varphi_n \in [0, 2\pi]$ is measured counterclockwise from the x axis, as shown in Figure 1. For simplicity, only normal wave incidence with respect to the z axis is considered in the following discussions.
With suitable data segmentation and Fourier transform, the frequency response of the \( M \times 3 \) complex array data snapshot vector is given by [4]

\[
\begin{align*}
Y_p(f_i) &= F B_p A(\phi) S(f_i) + N_p(f_i) \\
Y_v(f_i) &= F B_v A(\phi) S(f_i) + N_v(f_i) \\
Y_{vp}(f_i) &= F B_{vp} A(\phi) S(f_i) + N_{vp}(f_i)
\end{align*}
\]

\( i = 1, 2, \ldots, J \)

the matrices are defined as below,

\[
Y_p(f_i) = [Y_{p1}(f_i), \ldots, Y_{pm}(f_i)]^T, \quad Y_v(f_i) = [Y_{v1}(f_i), \ldots, Y_{vn}(f_i)]^T,
\]

\[
N_p(f_i) = [N_{p1}(f_i), \ldots, N_{pmb}(f_i)]^T, \quad N_v(f_i) = [N_{v1}(f_i), \ldots, N_{vmb}(f_i)]^T,
\]

\[
F = [w_{-K}, w_{-K+1}, \ldots, w_K],
\]

\[
w_q = [1, \exp(-j2\pi q/M), \ldots, \exp(-j2\pi q(M-1))], \quad A(\phi) = [a(\phi_1), \ldots, a(\phi_K)]
\]

\[
a(\phi_q) = [\exp(-jKq\phi_1), \ldots, \exp(jKq\phi_K)]^T, \quad B_p = \text{diag}[b_{-K}, \ldots, b_K], b_q = j^q J_q(k, r),
\]

\[
B_{vp} = \text{diag}[-Kb_{-K} / rk, \ldots, Kb_K / rk], \quad B_{vr} = \text{diag}[b_{-K}^*/jk, \ldots, b_K^*/jk]
\]

\( b_q \) is the derivative from \( b_a \) with respect to \( r, K = [k, r], k_i = 2\pi f_i / c \).

The transformation matrix \( T_p(f_i), T_v(f_i), T_{vp}(f_i) \) is defined as,

\[
T_p(f_i) = \frac{1}{M} B_p^{-1} F^H, \quad T_v(f_i) = \frac{1}{M} B_v^{-1} F^H, \quad T_{vp}(f_i) = \frac{1}{M} B_{vp}^{-1} F^H
\]

Transforming the actual array vector \( Y_p(f_i), Y_v(f_i), Y_{vp}(f_i) \) from element space into modal space

\[
\begin{align*}
Y_{pe}(f_i) &= T_p(f_i) Y_p(f_i) = A(\phi) S(f_i) + T_p(f_i) N_p(f_i) \\
Y_{ve}(f_i) &= T_v(f_i) Y_v(f_i) = A(\phi) S(f_i) + T_v(f_i) N_v(f_i) \\
Y_{vpe}(f_i) &= T_{vp}(f_i) Y_{vp}(f_i) = A(\phi) S(f_i) + T_{vp}(f_i) N_{vp}(f_i)
\end{align*}
\]

Assume when using a fixed order \( K \) for the broadband signal, the response of the beamformer may be made approximately constant over the design band.

Assuming the source signals and the noise are uncorrelated, the data covariance matrix is

\[
R_{pe}(f_i) = E[Y_{pe}(f_i) Y_{pe}^H(f_i) + Y_{ve}(f_i) Y_{ve}^H(f_i)]
\]

A broadband modal space data covariance matrix can be formed as

\[
R_{vp} = \sum_{j=1}^{J} R_{vp}(f_i)
\]
The modal space data covariance matrix is now in a form in which conventional eigen-based DOA estimators may be applied. For the MUSIC algorithm, the source directions are given by the $N$ peak positions of the following spatial spectrum [5]

$$P_{evo, music} = \frac{1}{a^H(\phi)U_nU_n^H a(\phi)}$$

(6)

Where $U_n$ is the eigenvectors corresponding to the smallest $2K+1 - N$ eigenvalues, $a(\phi) = [\exp(-jK\phi) \cdots \exp(jK\phi)]^T$ is the transformed steering vector in modal space.

3. SIMULATIONS AND EXPERIMENTAL RESULTS

The simulations are done with a UCAVSA of 8 elements. The radius $r=0.30m$, the fluid is water with $c=1500m/s$ and $\rho=1000kg/m^3$.

A set of simulations were performed to compare the performance between UCPSA and UCAVSA. Figure 2 shows the outputs of MUSIC algorithm for the UCAVSA and the UCPSA with different SNR. The incident far field broadband source originates from $\theta_0=90^\circ$ and $\phi_0=167^\circ$. Figure 3 shows the comparison of the average RMSEs for UCPSA and UCAVSA for several SNR. Figure 4 shows the comparison of the resolution performance for UCPSA and UCAVSA for several SNR.

(a)SNR =0dB  
(b) SNR =-20dB

Figure 2 Normalized output spatial spectrum for UCPSA and UCAVSA
The experiment of the application for UCAVSA was carried out in Song Hua Lake in Jilin province. In the experiment, the circular acoustic vector-sensor array was placed at the depth of about 15 m, with the distance from the mother ship about 200m. The depth of the lake at the experiment site was about 30m. The digitized signals were continuously recorded by the underwater acoustic multi-channel automatic measurement system. Figure 5 shows the target tracking results of the UCAVSA and UCPSA.
4. CONCLUSION

An approach to Broadband direction of arrival (DOA) estimation for a uniform circular acoustic Vector-Sensor Array is proposed. Simulation results show that the UCAVSA has distinct advantages over the same manifold of traditional uniform circular pressure-sensor array (UCPSA) in anti-noise performance, resolution and accuracy. The method is based on the principle of coherency between pressure and particle velocity, which can suppress interference in isotropic noise field. The algorithm could combine the anti-noise performance of the Vector-Sensor with spatial resolution performance of array system to solve the problem of high-resolution DOA estimation of remote targets for UCAVSA.

REFERENCES

A ROBUST NOISE SOURCES LOCALIZATION AND IDENTIFICATION METHOD BASED ON VECTOR SENSOR ARRAY

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Abstract: minimum variance distortionless response (MVDR) focused beamformer based on vector sensor array is widely used in the area of the noise source high-resolution localization and identification. However, when an arbitrary unknown signal steering vector mismatch occurs or the training sample size is small, the performance of MVDR focused beamformer will be severely degraded. In this paper, we develop a new approach, which is based on the worst-case concept, to improve the robustness of the original method. It is shown that the proposed algorithm improves its robustness by imposing the array response constraint on the uncertainty set of the steering vector, and can be reformulated in a convex form as the so called second order cone program (SOCP), and then solved efficiently using the well established optimization tool, Sedumi. Theory analysis and computer simulations show better performance of our robust beamformer as compared with the existing methods: it can achieve a greater dynamic range, sharper focused peak, and lower background noise level. The results in this paper demonstrate our proposed method can be applied in the underwater noise source high-resolution localization and identification.

Keywords: vector sensor array; robust; MVDR; focused beamformer; noise source localization and identification; high-resolution
1. INTRODUCTION

Focused beamforming (Near-field beamforming) is an effective technique to image sound sources using near-field measurements of the sound pressure and vector field, which can provide both intensity and position information [1][2]. Particularly, the minimum variance distortionless response (MVDR) focused beamformer based on vector sensor array, which can provide high-resolution and noise suppression performance, is widely used in the area of the noise source localization and identification[3-4].

When MVDR focused beamformer is applied to practical problem, the performance degradation of such adaptive beamformer may become even more pronounced than in the ideal case [5]. Many approaches have been proposed during the past three decades to improve the robustness of the MVDR beamformer. Take account of the array steering vector errors, additional linear constraints, including point and derivative constraints, can be imposed to improve the robustness of the MVDR beamformer [6]. The diagonal loading approaches has been a popular approach to improve the robustness of the MVDR beamformer [7]. Based on the optimization of worst-case performance, a new powerful approach to robust MVDR beamformer in the presence of an arbitrary unknown steering vector mismatch is proposed. The formulation of this approach involves minimization of a quadratic function subject to infinitely many nonconvex quadratic constraints, and can be reformulated as a convex second-order cone program (SOCP) and solved efficiently [8-9]. According the near-field measurement condition, the vector sensor array can be treated as a special arbitrary array because of the spherical wave curvature, and optimization of worst-case performance can also used to improve the performance of MVDR focused beamformer based on vector sensor array.

In this paper, Some background of MVDR focused beamformer is presented in Section 1, where several popular robust MVDR beamforming techniques are overviewed. In Section 2, we obtain the formulation of conventional and MVDR focused beamformer based on vector sensor array. In Section 3, we describe a robust focused MVDR beamformer based on the optimization of worst-case performance. Then, we convert it to a convex SOCP that can be efficiently solved. Section 4 presents some simulation results where the performance of the proposed method is compared with conventional and MVDR focused beamformer in same disturb situation. Section 5 contains our concluding remarks.

2. FOCUSED BEAMFORMER BASED ON VECTOR SENSOR ARRAY

![Fig.1 The near-field localization model](image-url)
The wave propagation from the sound sources and the vector measurement geometry are shown in Fig. 1. By spherical wave assumption, the $M \times 1$ dimension pressure array focused steering vector is represented as [10]:

$$\hat{\mathbf{A}}^{(p)}(\hat{\mathbf{r}}) = \left[ \frac{1}{\hat{r}_1} e^{-j \frac{2\pi}{\lambda} x_1}, \ldots, \frac{1}{\hat{r}_m} e^{-j \frac{2\pi}{\lambda} x_m}, \ldots, \frac{1}{\hat{r}_M} e^{-j \frac{2\pi}{\lambda} x_M} \right]^T$$

(1)

where $\lambda$ is wave number, $M$ is the number of array element, $\hat{\mathbf{r}}$ is the $M \times 1$ dimension distance vector, $\hat{r}_m$ ($m = 1, 2, \ldots, M$) represents the distance from the source to the $m$th array element.

Further more, the $4M \times 1$ dimension vector sensor array focused steering vector can be represented as:

$$\hat{\mathbf{A}}^{(v)}(\hat{\mathbf{r}}, \hat{\theta}, \hat{\phi}) = \left[ \hat{\mathbf{A}}^{(p)}(\hat{\mathbf{r}}) \right]^T \left( \hat{\mathbf{A}}^{(\phi)}(\hat{\mathbf{r}}, \hat{\theta}, \hat{\phi}) \right)^T \left( \hat{\mathbf{A}}^{(\phi)}(\hat{\mathbf{r}}, \hat{\theta}, \hat{\phi}) \right)^T \left( \hat{\mathbf{A}}^{(\phi)}(\hat{\mathbf{r}}, \hat{\theta}, \hat{\phi}) \right)^T$$

(3)

The vector sensor array focused steering vector $\hat{\mathbf{A}}^{(v)}(\hat{\mathbf{r}}, \hat{\theta}, \hat{\phi})$ is consist of pressure sensor array focused steering vector $\hat{\mathbf{A}}^{(p)}(\hat{\mathbf{r}})$ and the particle velocity sensor array focused steering vectors $\hat{\mathbf{A}}^{(\phi)}(\hat{\mathbf{r}}, \hat{\theta}, \hat{\phi})$ and $\hat{\mathbf{A}}^{(\phi)}(\hat{\mathbf{r}}, \hat{\theta}, \hat{\phi})$. The $\hat{\theta}$ and $\hat{\phi}$ are the angle polar vector and azimuthal vector respectively.

The conventional and MVDR focused beamformer based on vector sensor array (VCFB and VMVDRFB) can be obtained as:

$$P^{(v)}_{\text{VCFB}}(\hat{\mathbf{r}}, \hat{\theta}, \hat{\phi}) = \left( \hat{\mathbf{A}}^{(v)}(\hat{\mathbf{r}}, \hat{\theta}, \hat{\phi}) \right)^H \mathbf{R}^{(v)} \hat{\mathbf{A}}^{(v)}(\hat{\mathbf{r}}, \hat{\theta}, \hat{\phi})$$

(4)

$$P^{(v)}_{\text{VMVDRF}}(\hat{\mathbf{r}}, \hat{\theta}, \hat{\phi}) = \frac{1}{\left( \hat{\mathbf{A}}^{(v)}(\hat{\mathbf{r}}, \hat{\theta}, \hat{\phi}) \right)^H \left( \mathbf{R}^{(v)} \right)^{-1} \left( \hat{\mathbf{A}}^{(v)}(\hat{\mathbf{r}}, \hat{\theta}, \hat{\phi}) \right)}$$

(5)

where $\mathbf{R}^{(v)}$ is the $4M \times 4M$ dimension data covariance matrix.

3. ROBUST FOCUSED BEAMFORMER BASED ON WORST-CASE PERFORMANCE OPTIMIZATION

In this section, we develop a new focused beamformer based on vector sensor array that is robust against an arbitrary signal steering vector mismatch and small training sample size. Our approach is based on the worst-case performance optimization. We assume that the norm of the steering vector distortion $\Delta$ can be bounded by some known constant
\( \varepsilon > 0 \), \( \| \Delta \| \leq \varepsilon \). Then, the actual focused steering vector \( \hat{A}^{(v)}(\hat{r}, \hat{\theta}, \hat{\phi}) \) belongs to the set \( A(\varepsilon) = \{ A^{(v)}(\hat{r}, \hat{\theta}, \hat{\phi}) | A^{(v)}(\hat{r}, \hat{\theta}, \hat{\phi}) = \hat{A}^{(v)}(\hat{r}, \hat{\theta}, \hat{\phi}) + \Delta, \| \Delta \| \leq \varepsilon \} \).

The absolute value of the array response should not be smaller than one.

\[
|wA^{(v)}(\hat{r}, \hat{\theta}, \hat{\phi})| \geq 1, \quad A^{(v)}(\hat{r}, \hat{\theta}, \hat{\phi}) \in A(\varepsilon) \tag{6}
\]

The robust formulation of adaptive beamformer can be written as the following constrained minimization problem:

\[
\begin{align*}
\min_{w} & \quad w^H R^{(v)} w \\
\text{s.t.} & \quad |wA^{(v)}(\hat{r}, \hat{\theta}, \hat{\phi})| \geq 1 \quad \text{for all} \quad A^{(v)}(\hat{r}, \hat{\theta}, \hat{\phi}) \in A(\varepsilon)
\end{align*} \tag{7}
\]

Further more, this problem can be rewritten as:

\[
\begin{align*}
\min_{w} & \quad w^H R^{(v)} w \\
\text{s.t.} & \quad w^H A^{(v)}(\hat{r}, \hat{\theta}, \hat{\phi}) \geq \varepsilon \| w \| + 1, \quad \text{Im}\left\{ w^H A^{(v)}(\hat{r}, \hat{\theta}, \hat{\phi}) \right\} = 0.
\end{align*} \tag{8}
\]

Note that the problem (8) has much simpler formulation than (7) and is convex. The next step involves developing a SOC formulation of (8), we convert the quadratic objective function of (8) to a linear one. Let \( R^{(v)} = U^H U \) be the Cholesky factorization of \( R^{(v)} \). We can convert the objective function of (8) into \( w^H R^{(v)} w = \| U w \|^2 \). Apparently, minimizing \( \| U w \| \) is equivalent to minimizing \( w^H R^{(v)} w \). Hence, introducing a new scalar non-negative \( \tau \) and a new constant \( \| U w \| \leq \tau \), we can convert (8) into the following problem:

\[
\begin{align*}
\min_{\tau, w} & \quad \tau \\
\text{s.t.} & \quad \| U w \| \leq \tau, \quad \varepsilon \| w \| \leq w^H A^{(v)}(\hat{r}, \hat{\theta}, \hat{\phi}) - 1, \quad \text{Im}\left\{ w^H A^{(v)}(\hat{r}, \hat{\theta}, \hat{\phi}) \right\} = 0
\end{align*} \tag{9}
\]

Introducing, \( w = \Re\{ w \}^T, \quad \Im\{ w \}^T \),

\[
\begin{align*}
\tilde{A} &= \left[ \Re\{ \begin{pmatrix} A^{(v)}(\hat{r}, \hat{\theta}, \hat{\phi}) \end{pmatrix} \}^T, \quad \Im\{ \begin{pmatrix} A^{(v)}(\hat{r}, \hat{\theta}, \hat{\phi}) \end{pmatrix} \}^T \right]^T, \\
\tilde{A} &= \left[ \Im\{ \begin{pmatrix} A^{(v)}(\hat{r}, \hat{\theta}, \hat{\phi}) \end{pmatrix} \}^T, \quad -\Re\{ \begin{pmatrix} A^{(v)}(\hat{r}, \hat{\theta}, \hat{\phi}) \end{pmatrix} \}^T \right]^T, \quad \tilde{U} = \begin{bmatrix} \Re\{ U \} & -\Im\{ U \} \\ \Im\{ U \} & \Re\{ U \} \end{bmatrix}
\end{align*}
\]

We rewrite (9) in terms of real-valued vectors and matrices as

\[
\begin{align*}
\min_{\tau, w} & \quad \tau \\
\text{s.t.} & \quad \| \tilde{U} w \| \leq \tau, \quad \varepsilon \| w \| \leq \tilde{w}^T \tilde{A} - 1, \quad \tilde{w}^T \tilde{A} = 0
\end{align*} \tag{10}
\]

The weight vector of the robust focused beamformer based on vector sensor array is given by \( w_{\text{opt}} = [\tilde{w}_1, \ldots, \tilde{w}_M]^T + j[\tilde{w}_{M+1}, \ldots, \tilde{w}_{2M}]^T \). The robust focused beamformer (7) is converted to the canonical SOC problem which can be easily solved by using standard and highly efficient interior point method software tools, such as SeDuMi convex optimization MATLAB toolbox[11].
4. SIMULATION

Defined steering vector disturb parameter as:

\[
\text{Disturb} = 10 \log_{10} \left( \frac{\| \delta \|_2^2}{\| \delta^{\text{true}} \|_2^2} \right)
\]  

(11)

The arbitrary unknown signal steering vector mismatch can be described by this parameter. In our simulation, we assume a uniform linear vector sensor array with \(M=9\) sensors spaced half a wavelength apart. The sound frequency and data sample rate are \(f=1kHz\) and \(f_s=32.768kHz\) respectively, and the sound speed is 1500m/s. According the sound position coordinate is \((x_s,z_s)=(2,2)m\) and distance \(y_s=3m\), the scanning region are -5m~5m in x direction and -5m~5m in z direction. The snapshot is 2048, and signal-to-noise ratio (SNR) and disturb parameter are 10dB and -10dB. In our simulation results, the performance of the robust focused beamformer (RMVDRFBF) is compared with the both conventional and MVDR focused beamformer (CFBF and MVDRFBF). The compared results are shown in Fig.2 and Fig.3 \(\varepsilon=0.5\).

![Spatial spectrum with the same disturb](image1)

(a) CFBF  (b) MVDRFBF  (c) RMVDRFBF

**Fig.2 Spatial spectrum with the same disturb**

![Spatial spectrum slices in direction](image2)

(a) Spatial spectrum in direction x  (b) Spatial spectrum in direction z

**Fig.3 Spatial spectrum slices in direction x and z**

The simulation figures clearly demonstrate that the RMVDRFBF consistently enjoys the best performance among the three methods compared. In the same disturb and mismatch situation, the CFBF has much poorer resolution than MVDR focused beamformer, and the sidelobes of which gives false peaks. However, the MVDRFBF has the performance degradation. Compared to the above methods, the RMVDRFBF can achieve a greater dynamic range, sharper focused peak, and lower back-ground noise level. The results in this paper demonstrate the robust focused beamformer can be applied in the underwater noise source high-resolution localization and identification.
5. CONCLUSION

In our discussion in this paper, we have shown how to obtain a robust MVDR focused beamformer based on the worst-case performance optimization. A convex formulation for such a robust MVDR focused beamformer problem is derived using second-order cone programming. It is shown that the robust beamformer has better performance, such as greater dynamic range, sharper focused peak, and lower background noise level. The results in this paper demonstrate our proposed method can be applied in the underwater noise source high-resolution localization and identification.

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Bioacoustics
Bioacoustic Absorption Spectroscopy of Physoclists

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Abstract: This paper describes the results of a recent Bioacoustic Absorption Spectroscopy experiment, which was focused on a physoclist, North Pacific hake, Merluccius productus. Physoclists are a class of fish that can control the amount of gas in their swim bladders. The experiment included coincident measurements of transmission loss (TL) vs. frequency (f) at 0.3 < f < 5 kHz, fish layer depths, fish length distributions, and continuous temperature profiles. TL measurements were conducted several times between moving, ship-deployed, broadband sources and a 24 element vertical array at ranges between 0.1 and 10 km. Measured resonance frequencies, which were attributed to hake, decreased with time during the day, and approached values, which were consistent with theoretical calculations of fully filled swim bladders at night. These results are in accord with previously reported laboratory measurements of the temporal evolution of the resonance frequencies of physoclists.

Keywords: bioacoustics, absorption, spectroscopy, resonance frequency, physoclist, hake
INTRODUCTION

Previously reported Bioacoustic Absorption Spectroscopy (BAS) experiments were conducted on physostomes (sardines and anchovies), a class of fish that cannot control the volume of gas in their swim bladders (Diachok, 1999, Diachok and Wales, 2005). The volumes of their swim bladders change instantaneously with depth in accord with Boyle’s Law. The objective of this paper is to report initial results from an experiment designed to investigate bioacoustic absorptivity by a physoclist, North Pacific hake, Merluccius productus. Physoclist are a class of fish that can control the amount of gas in their swim bladders. Laboratory measurements, reported by Tytler and Blaxter (1973), indicate that following an increase of pressure by a factor of two, physoclists require ~24 hours to fill their swim bladders; whereas following a decrease in pressure by a factor of two requires only ~3 hr to resorb excess gases. Based on these results, Tytler and Blaxter postulate that vertically migrating physoclists are rarely adapted during daytime, and are most likely adapted only at night. Echo sounder measurements reveal that hake undergo diurnal vertical migration on the continental shelf (Alverson and Larkin, 1969). Consequently, measurements of the resonance frequencies of hake in the vicinity of the shelf break may be expected to exhibit evidence of adaptation.

THE EXPERIMENT

Bioacoustic Absorption Spectroscopy (BAS) measurements were conducted from the RV New Horizon in August 2012 in the vicinity of the shelf break at 42.5 N, at a mesoscale (~20 km diameter) biological hot spot where the concentration of hake was determined to be relatively high. The location of this hot spot was derived from echo sounder and trawl measurements, which were conducted by the Northwest Fisheries Science Center (NWFSC) 10 days before the BAS experiment. The location of this hot spot was confirmed with an echo sounder survey, which was conducted from the RV New Horizon one day before the experiment.

TL measurements were conducted along 10 km tracks, which were approximately north and south of the array, parallel to shore, about 260 m deep and relatively flat. The configuration of the BAS experiment is illustrated in Figure 1. Low frequency (0.3-1.5 kHz) and mid-frequency (1.5-5 kHz) broadband sources were towed at a depth of about 95 m (Gauss et al., 2009). These sources were programmed to transmit sixty CW tones between 220 Hz and 5 kHz (220, 233, 247…Hz) \( (f_{n+1} =1.059 \ f_n) \) over about 10 minutes, every 10 minutes. The source level was < 180 dB at all frequencies. The duty cycle was 10 %. Signals were received on a vertical hydrophone array that spanned the water column. The full array consisted of 6 EARS arrays (Lammers et al., 2008). Each EARS array consisted of 4 hydrophones separated by 4 m, and included a digital recorder. The mid points of the EARS arrays were at 30, 54, 114, 164, 226 and 246 m. Measurements of TL vs. range and frequency were conducted successfully 3 times at night, and 4 times during the day. The transit speed was 3 kt. Thermistor strings were deployed...
along the receiving array and at the ends of the measurement tracks, to permit measurement of the temporal evolution of average sound speed profiles along the propagation path. Temperature profiles did not change significantly with time of day. The bottom at this site consists of mud. The attenuation coefficient due to mud in the frequency range of interest is small (Diachok and Wales, 2005).

Depths of fish and plankton layers were measured with an 18 kHz echo sounder. The resultant data, illustrated in Figure 1, suggest that fish (probably hake) were present near the bottom during the day and to a lesser extent at night. High concentrations of fish (possibly hake) were evident at ~30 m and ~90 m at night. Fish evident near the surface during the day were hypothetically *myctophids*. Sunrise and sunset on August 3 occurred at 0557 and 2036 respectively. Vertical migrations of fish (hypothetically hake) were evident at dawn at ~0700 and at dusk at ~2100 respectively. Inferences of layer depths of hake were guided by the requirement for consistency between echo sounder and TL data. Theoretical calculations indicate that peak values of bio-attenuation coefficients coincide with depths of fish layers (Diachok, 1999). Inversion of layer depths is beyond the scope of this paper. Trawls conducted by the NWFSC indicated that the dominant species near the bottom in the vicinity of the shelf break during daytime were 31 cm hake. Trawls conducted by the Southwest Fisheries Science Center (SWFSC) indicated that 4 cm long *Tarletonbeania crenularis*, a myctophid, was the dominant species in the vicinity of the shelf break near the surface at night. SWFSC trawls also indicated the presence of sardines and anchovies on the continental shelf at water depths less than 100 m.

Figure 1. Measurement geometry for TL measurements between towed source and vertical array, and echo sounder records at ~0100 at night (left) and ~1900 during the day (right).

**RESULTS**

The attenuation coefficient, A, vs. frequency and depth was calculated by correcting received levels for cylindrical spreading loss and chemical absorption loss. Figure 2 provides a contour
plot of A at ~1900 L during the day and 0100 L at night. The peaks in frequency-depth space at
night at 1 kHz at d < 60 m and at 1.7 kHz at ~226 m are consistent with adapted 31 cm long hake
(h). The peak at night at ~3.3 kHz at d < 30 m is attributed to 4 cm long myctophids (m). The
peak at ~2.5 kHz at ~246 m near the bottom during the day is attributed to non-adapted hake, and
the peak at 3 kHz at d < 30 is attributed to myctophids. The magnitude of A attributed to hake is
higher during the day when hake concentrate in one layer near the bottom. The resonance
frequency attributed to hake during the day decreased with time from ~4.3 kHz at 0800 L to ~2.5
kHz at 1900 L, and approached frequencies of adapted hake at night, 1.7 kHz at 0100 L, as
illustrated in Figure 3 (left). The cause of the peak at ~4 kHz near the bottom at night is unknown.
Analysis of the resonance frequencies attributed to myctophids is beyond the scope of this paper.

Figure 2. Attenuation coefficient (dB/km) vs. frequency and receiver depth, derived from TL
measurements at ~0100 L at night (left) and ~1900 L during the day (right).

Figure 3 (left) also shows theoretical calculations, in accord with

\[ f = 322 \varepsilon (1 + 0.1 d)^n / r \]  

where r (in cm) is the effective radius of the swim bladder at the surface, \( \varepsilon \) (non-dimensional) is
the correction for the eccentricity of the swim bladder, \( d \) is the depth (in meters), and \( n = 1/2 \) for
physoclists (heavy curve) and 5/6 for physostomes. The calculations shown in this figure assume
that \( r \) equals 1.0 cm for physoclists and 1.2 cm for physostomes. In both cases \( \varepsilon \) was assumed to
equal 1.2. The measurements shown in this figure suggest the following sequence of events: the
hake, which reside near the surface at night, descend to about 200 m at dawn, changing
frequencies in accord with Boyle’s Law, and then fill their swim bladders during the day,
approaching resonance frequencies of fully filled swim bladders at night. This sequence of
events resembles Lovik and Hovem’s (1979) laboratory measurements of the evolution of
resonance frequency of a physoclist. Figure 3 (right) shows that the resonance frequencies of an
8 cm long coalfish, which was adapted at 6 m and instantaneously transferred to 32 m, initially
increased frequencies in accord with Boyle’s Law; and then decreased and approached the
resonance frequencies of fully filled swim bladders in 24 hours. The data shown in Figure 3 (left) suggests that hake fill their swim bladders in ~18 hours, following migration from 30 to 200 m. It is noteworthy that Dezhang Chu’s (unpublished) measurement of the resonance frequency of hake, which were derived from broadband echo sounder data, 2.5 kHz at 1500 L at 120 m, also suggested that hake swim bladders were not adapted during daytime.

The data shown in Figure 3 (left) suggests that the value of r of 31 cm hake at the surface equals 1.1 ± 0.1 cm, which is in good agreement with Henderson and Horn’s (2007) laboratory measurements, as illustrated in Figure 4. It is noteworthy that Nero et al.’s (1998) estimate of r of 45 cm long hake (1.7 cm), which was derived from back-scattering measurements in deep water, is also consistent with the laboratory data.

![Figure 3. Resonance frequencies of 31 cm hake from BAS measurements (left) and 8 cm coalfish from laboratory measurements (LoviK and Hovem, 1979) (right) and theoretical calculations.](image)

![Figure 4. Effective radius of adapted hake vs. length derived from BAS (o) and laboratory measurements (Henderson and Horn, 2007).](image)
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EFFECTS OF OFFSHORE WIND FARMS OPERATIONAL NOISE ON BLUEFIN TUNA BEHAVIOUR

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\textbf{Abstract:} The number of offshore wind farms is growing up quickly in the lasts years. Several studies about its environmental acoustic impacts have been developed at the same time the industry expands, most of them related to the high level impulsive noise produced during the pile diving process associated to the construction stage. Nevertheless, the study of the impact of the operational noise of turbines is very limited. In this paper we investigate the behavioural response of Bluefin tuna when exposed to the operation noise of a turbine. We analysed tuna reaction in terms of three parameters: depth of the school, swimming pattern and changes in the swimming direction. The experiment was developed in a fixed commercial tuna cage in the Mediterranean Sea. The usual behaviour of Bluefin tuna in captivity conditions was previously analysed using a continuous monitorization. Variations in depth were observed when feeding boat approaches, which could be interpreted as a consequence of the acoustical stimulus. The turbine noise was acoustically characterized, and reproduced using a broad-band underwater source. To monitor tuna behaviour two echosounders and a video system were simultaneously used. When exposed to short duration noise tuna behaviour does not exhibit clear disturbances. Nevertheless, with long duration emission tuna reacted: school reduced the radio of the circular swimming region, moved up to the surface and some individuals were disorientated. Tuna seems to be habituated after several repetitions is short time.

\textbf{Keywords:} Offshore wind farm, environmental impact, Bluefin tuna
1. INTRODUCTION

Offshore wind farms are one of the most promising power sources nowadays, and they are suffering a fast expansion along the coastal areas. Nevertheless, the environmental impacts regarding to the construction and operation of this infrastructures have not been yet completely evaluated, and several studies have been focus on that problem during last years [1]. From the underwater acoustical point of view, most of the studies regard the impact of high intensity pile driving noise produced during the construction period on biological enviroment: benthos, bivalves, marine mammals or fishes, with a wide variety of results, from no evidence of injury or reaction to the immediate death [2,3 and references therein]. Even less is the number of papers concerning the impacts of the operational noise of wind farms, and they consider only a reduced number of target species [Thomsen and therein]. The aim of this work is to contribute to the knowledge of the potential effects of operational noise of wind turbines on the behaviour of Bluefin tuna (*Thunnus thynnus*). Although dramatic consequences as death or physical damage are not expected in this case, the continuous noise associated to the low level long-term operation regime of turbines might affect the behaviour of fishes, which could interfere with feeding processes and migration routes.

Bluefin tuna is a high economic resource and it has been matter of study during lasts years due to the decrease of the stock and the limitation and the control on fishing quotas. In spite of the high economic impact of this specie, the number of studies regarding the effect of underwater sources of noise on them is very limited [4] and only recently the characterization of hearing threshold of similar specie Bluefin tuna (*Thunnus orientalis*) has been dealt with [5]. Tuna form schools that migrate at ocean scales across the Gibraltar strait from Atlantic Ocean to Mediterranean Sea, where their migration routes pass nearby the coastal regions.

In order to investigate the reaction of tuna to the operational turbine noise, Bluefin tuna located in a fixed commercial fattening cage in the Mediterranean Sea were exposed to a noise equivalent to the operational wind turbine noise previously recorded. The animals were continuously monitorized during weeks before the sound exposure using an acoustical and visual system, to ensure that the reactions that could be observed after noise emission were distinguishable from the usual behaviour of the fishes. The behaviour was characterized in terms to three variables: position of fish school along the water column, swimming pattern and changes in swimming direction. As was expected tuna school exhibit different

2. MATERIAL AND METHODS

The acoustical recording system consists of a scientific Biosonics DT-X echosounder, working at 201 kHz and insonificating an angle of 22.5 ° at -3 dB. The transducer was located looking up at the surface at 28 meters deep at the bottom of a floating commercial cage of 50 m of diameter, in the middle of one of the radius of the pen. The fattening cage was located in the Mediterranean Sea, in front of L’Ametlla de Mar village at 4000 m from the coast. Together to the transducer was located a video camera protected in a watertight box. During the continuous monitoring study, the system was feeding by external batteries hanging on the cage inside a waterproof box, together with the
electronics for controlling the system, data transfer was done by a wifi communication and system was remotely controlled. Tuna school behaviour was continuously monitored along six weeks. During the exposition to the turbine noise, an extra single-beam Knudsen echosounder was installed in the cage to control the part of the school that was out of the region insonificated by the other echosounder to ensure that the distribution and behaviour of tuna has not evident differences along the cage. The video camera was located alongside the measurement echosounder. A scheme of the system is shown in Figure 1.

![Scheme of the experimental setup](image)


The experiment was designed to test the effect of operational turbine noise on Bluefin tuna behaviour. The noise of a wind turbine was previously recorded 50 meters from the source during 30 seconds and sampled at 350 kHz. The wind farm turbine produces a broad band noise (from 40 to 10000 HZ) with different characteristic peaks at 50 Hz (∼142 dB ref 1 μPa) at 50 meters from the source. A broad-band Data Physics GW350 underwater source property of the Spanish Army was used to reproduce the turbine sound.

3. RESULTS

3.1. Usual behaviour

The behaviour of tuna school was continuously monitored along six weeks during January and February of 2013. The result of monitorization consisted in an amount of 700 hours of acoustic recordings and 150 hours of video. Data were analyzed using software developed specifically for this purpose in Matlab® code.

Tuna school usually swims in a circular pattern covering a large area of the cage, swimming closer to the cage nets. As expected [6], school depth exhibited day/night variations. During the middle of the day the school tends to be closer to the surface, going deeper overnight. This behaviour was observed repeatedly during the period of continuous monitoring, recording an average difference $\Delta = 2.3$ meters between day and night depths.

The school also reacts to the feeding boat. Tunas were fed with frozen mackerel blocks were thrown through a tube from the boat and floats in the middle of the cage. When feeding boat arrived and moored beside the cage, before the food was launched, tuna escape from the surface and swim deeper. The school remains far from the surface until the boat departs and then rises up again. Fig. 2 shows the echograms corresponding to the described process, where the distance from cage bottom, $d$, is indicated. This behaviour
can be interpreted as related to the feeding boat noise, well as an avoidance movement, well as a feeding manoeuvre. In any case, it seems to be clear that, as expected [4], tuna react to noise, and so they can be affected by the turbine noise we want to test.

Fig. 2. Echograms corresponding to feeding process at 15-01-2013. (a) Boat approaches (dashed line), $d_i=11.6m$ and $d_{ii}=9.17 m$, (b) Boat moored unloading food block. Tuna going deeper $d=9.8m$, (c) Food block through the acoustic beam. School still far from the surface $d=10.1m$, (d) boat departs from the cage and tuna rise again $d=12.3m$.

3.2. Effect of turbine noise

The noise produced by a turbine was recorded at 50 meters from the source, sampled at 350 KHz. The noise was reproduced using a previously characterized broad-band Data Physics GW350 underwater source. The sound was reproduced, as shown in Fig. 3.

Fig. 3. Original recorded turbine emission (black) and reproduced noise used in the experiment (red).

To ensure the health of animals, the experiment was limited to not exceed the usual acoustic levels to which they were subjected tuna. In order to satisfy this requirement, the acoustic environment of the cage was recorded and the maximum level registered, corresponding to the shot of bang stick when tuna are sacrificed determined the maximum sound level of turbine noise emission (maximum SPL ~165 dB ref $1\mu Pa$), which corresponds to a turbine working at 6.5 meters. Reaction under short (from ten to fifteen seconds) and
long (from ten to fifteen minutes) stimuli were tested. Results can be summarized as follows (Fig.4):

i) Short time emission. First time that tuna were exposed to the short time duration turbine noise, they moved fast to the surface, in a clear avoidance manoeuvre from the noise source. When sound ceased, they recovered the original distribution along the cage. Nevertheless, this behaviour could not be observed again, and tuna did not react any more time to this short time stimulus. It should be note that the time interval between different measurements was very short due to technical limitations.

ii) Long time emission. In this second case, tuna behaviour evidenced the reaction to turbine noise during and after the sound exposure.

- School depth: after some minutes from the beginning of noise emission, school moves up. Tuna remain swimming closer to the surface even when acoustic emission has finished, and only some minutes later recover the original distribution.

- Swim pattern: tuna bunched and swim closer together; they still swim like a school with a circular pattern, but with a smaller radius, and only occupy half part of the cage. It could be observed from the control boat and assessed by data from control echosounder and video camera: almost no tuna was recorded during sound emission

- Disorientation: during the minutes after emission, some specimens swam in opposite direction to the rest of school, which could be interpreted as slight disorientation. Several five minutes intervals were analysed before (ten random intervals) and after (one after any emission) the sound emission. During the intervals previous to the emission, any tuna changed his swimming direction from the school one, nevertheless after long time noise emission, an average of 15 tuna in 5 minutes were registered to swim in opposite direction with higher speed. (We note that we only could register the changes of swimming direction after noise emission, because tuna were out from the visual range of the camera).

![Echograms corresponding to ten minutes noise emission at 17:15h with different positions of school in water column](image)

**Fig.4.** Echograms corresponding to ten minutes noise emission at 17:15h with different positions of school in water column (a) $d=19.03$ m, (b) $d=21.43$ m and (c) $d=20.13$ m ($d$ measured from bottom).

The experiment was developed in absence of shipping in the neighbourhood of the cage (background SPL 110/120 dB ref $1\mu$Pa). Second time that it was repeated, tuna exhibited the same kind of behaviour, but with longer reaction time. Third time, and under noisier conditions, tuna behaviour did not exhibit evident variations, showing a high degree of adaptability to noise.
4. CONCLUSIONS

By exposing tuna to wind turbine low frequency noise, main reactions are shown to high levels and longtime exposures. These reactions can be summarized as: i) position change in the water column of the fish school, ii) contraction of the school (avoidance), iii) slight disorientation of some specimens and iv) increased speed.

This behavior was repeatedly observed with longtime emission in absence of other noise sources, and emission levels $\sim 165 \text{ dB}_{1 \mu \text{Pa}}$. By the behavior shown during short time exposure, tuna seems to exhibit a high degree of adaptability.

In spite that the study clearly shows the Bluefin tuna reaction to turbine noise, the implications of this behavior alteration is far to be still clear. The study was limited in time by economic reasons, but more exhaustive events should to be recorded and analysed for a complete characterization of effect of turbine noise on tuna. Finally, we cannot forget that we work with semicaptive animals which have not the same constraints that tuna in nature, they are limited in space and habituated to shipping traffic noise.

The effect of new human activities at ocean on ecosystems can only be measured more specific studies of affected species in each geographical area, improving the knowledge of physiology and behaviour, and developing methodologies and tools for studying and monitoring, and research in this line is needed.

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ACOUSTICAL BIOMASS ESTIMATION RESULTS IN MEDITERRANEAN AQUACULTURE SEA CAGES

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Abstract: Biomass estimation, both weight distribution and fish density, are clue factors for fish farming management. We present the experimental results of the acoustical estimates for both variables in Mediterranean intensive aquaculture floating cages. Sea bass and gilthead sea bream cages have been studied using scientific split beam echosounders with transducers placed at the sea surface. Dorsal measurements of the acoustical target strength (TS) of isolated fish tracks in the upper limit of the dense caged schools correlate with strong fish mean size differences. Mean acoustical volume backscattering strength monitoring reveals pen manipulations and fish stock changes, with good agreement in fish density estimations for the studied conditions.

Keywords: aquaculture, biomass estimation, echosounder, target strenght
1. MOTIVATION

In the last decades aquaculture has become a fundamental activity to cover the human needs of food, providing today the 50% of the products with aquatic origin, and having an increasing demand, supposed to reach the 65% in 2030 [1]. Fish farming constitutes almost the half of the aquaculture effort, and several of the most important commercial fish species, like Atlantic salmon (Salmo salar), gilthead sea bream (Sparus aurata) or sea bass (Dicentrarchus labrax) are intensively cultivated in floating sea cages, where juvenile stocks are introduced to be fed until they reach the desired commercial sizes. A critical aspect of the production process, from both the economical and the environmental impact point of view, is the adequate dosage of pelleted food, which is calculated as a function of the fishes biomass, in order to achieve optimal growing rates. Fish size monitoring tools are then crucial to improve the farms management, and different sampling methods have been developed, being the stereoscopic optical image recording the most successful non-invasive one, since it provides fish size measurements (length and height) which are the input for biometric relationships to obtain fish weight [2].

Nevertheless the estimation of total biomass in the cage remains as an open problem and the use of acoustics have been proposed to cover the limitations of optical techniques. Acoustical target strength (TS) measurements from single fish tracks for size monitoring and volume backscattering strength (Sv) scaling for fish density and total biomass estimations are the common proposed approaches in fisheries acoustics stock assessment [3]. TS measurements issues related to its application in aquaculture for growing monitoring were deeply investigated for Atlantic Salmon in [4], and their results were mostly replicated for gilthead sea bream in close range experiments [5], having as main conclusion that TS correlates with fish size only for the ventral aspect measurements, showing dorsal TS values of smallest fish size classes tri- and bi-modal distributions related to fish directivity and remaining mean TS values almost constant with size for measurements taken from the surface. Total biomass estimation following the principle of linear superposition of the backscattered energy was proposed following the ventral measurements scheme (from below the floating cages) and a research project financed by the Norwegian Research Council developed with partially negative results: Sv measurements followed the increasing biomass in a Salmon cage during a production cycle (from 100 g until 2 kg fish growing) until it collapsed when the fishes were around 500g, corresponding with the highest values of biomass density in the cage. Different beam extinction techniques were applied to compensate acoustical shadowing but it resulted in a higher measured biomass than expected, possible due to multiple/forward scattering and school behaviour effects [6]. To our knowledge no other initiatives have been reported in the literature on this subject. However, during the state-of-art vigilance phase (beginning of 2012) of the Spanish technological development project for total biomass estimation in off-shore fish cages (ARM/1790/010), impulsed by the Asociación Empresarial de Productores de Cultivos Marinos de España (APROMAR), led by the Andalusian Aquaculture Technology Centre, CTAQUA, and participated by the UPV as scientific advisor, a communication of the company Biosonics Inc. stated that the American company had developed a system for biomass monitoring in aquaculture, mainly tested in Salmon farming, with very good results in growing monitoring and promising expectations in total biomass estimation, both from an echosounding scheme from the cage surface. Taking into account the previous results and the company proposals
it was decided to re-visit the dorsal scheme and the evaluation of the approach with production conditions in Mediterranean fish farms.

2. MEASUREMENTS IN MEDITERRANEAN SEA CAGES PRODUCTION CONDITIONS

Gilthead sea bream and sea bass cages with different sizes and densities were measured in September of 2012 in the Grupo Culmarexs plant of Águilas (Murcia, Spain) by Biosonics Inc. engineers in order to evaluate the application of their technology to the estimation of fish growing and total biomass with these Mediterranean species of great commercial interest. The technical assessment was performed by commitment of CTAQUA with the participation of their technicians at the sea campaign. The technical report elaborated by Biosonics Inc. stated that TS measurements were consistent with the size of measured fishes and that S_v scaling showed a correlation with the biomass variations due to the fish farm commercial catches and stock shifts operations. The report concluded that new experiments were needed to obtain proper TS vs. size relationships for the Mediterranean species and to monitor S_v long-time series to identify the periods with minimum variation coefficient in the mean S_v, when the school was supposed to occupy the cage uniformly, and estimated fish density in the insonified volume could be extrapolated to all the cage. In the following, an analysis of the acoustical data obtained in such campaign, but performed by our group in order to validate the mentioned report is detailed, together with the evaluation of an additional measuring campaign performed by us in September 2013 in a commercial cage close to slaughter with decreasing values of gilthead sea bream total biomass.

2.1. Target strength analysis for size estimation

We analysed the acoustical data obtained for three size classes of sea bass (of weight means 60, 480, and 1390 g) and one of gilthead sea bream (433 g) with the help of Sonar-5 Pro software (www.lindem.net).

The used echosounder configurations are given in Table 1. In order to avoid shadowing effects we took into account isolated tracks above the dense school, with an average measuring distance between 3 and 5 m; some examples of fish traces can be seen in the amplitude echogram of Fig. 1. Unimodal TS distributions were found for all sizes, species
and frequencies (see Fig 2). The only TS vs. size adjustment possible (only three size classes of sea bass were available) offered excellent correlation coefficients as shown in Fig. 3. Note the slope deviation from Love expression [3].

<table>
<thead>
<tr>
<th>Frequency (kHz)</th>
<th>123</th>
<th>201</th>
</tr>
</thead>
<tbody>
<tr>
<td>Ping interval (ms)</td>
<td>200</td>
<td>200</td>
</tr>
<tr>
<td>Beam aperture (degrees)</td>
<td>7.5</td>
<td>6.6</td>
</tr>
<tr>
<td>Source level (re 1 mPa @ 1m)</td>
<td>220.5</td>
<td>222.8</td>
</tr>
<tr>
<td>Pulse duration (ms)</td>
<td>200</td>
<td>300</td>
</tr>
</tbody>
</table>

**Table 1: Echosounder configurations used in Águilas campaign at Culmarex group facilities by Biosonics Inc. engineers.**

![Fig 2: Uni-modal dorsal TS distributions obtained with 123 kHz (left) and 201 kHz (right) split beam analysis for increasing mean weights (60-484-1319 g) in sea bass cages.](image)

**2.2. Sv measurements for fish density and total biomass estimation**

The time evolution of Sv was studied in order to evaluate the possibility of obtaining fish density estimations consistent with farm data. Unfortunately, most of the biomass data for higher densities (corresponding to growing cages) were just estimations of the producers. These estimations can have deviations up to 25 or 30%.
The only reliable biomass data proceed from cages close to slaughter, known as "commercial" cages, with reduced densities obtained from partial catches from growing cages. In any case, the observation of the temporal behaviour of mean $S_v$ reveals a strong variability increased when the cage volume was modified by during partial catches. In absence of manual operations, the variation coefficient of the $S_v$ along the water column took minimum values during the night and first hours of the day, and the mean value of this variable has consistent variation with the biomass changes after catches or stock shifts between cages. Even more, Figure 4 shows the good agreement of acoustical density estimation during these time intervals of reduced coefficient variations for both operation frequencies with measurements taken in different days.

In order to validate these positive preliminary results for total biomass estimation we realised a second experiment in a commercial cage of gilt-head sea bream in September of 2013 in Piagua (Almeria). The measured mean size was in this case an only class around 541 g. The fishes were initially transferred to the commercial cage from a growing cage up to a biomass density of 6,2 kg/m³. Two consecutive catches in a week time reduced the density to 4,9 and 3,5 kg/m³. The mean $S_v$ was remotely monitored with an equivalent configuration but using a Simrad EK60 200kHz scientific echosounder. Figure 5 plots the time evolution of the variable for recordings corresponding to different reference densities.

3. CONCLUSIONS

Dorsal TS measurements in sea bass cages offer a good correlation with strong class size differences in production conditions, where size dispersion in the cage can play a
significant role. Additional measurements are necessary to establish weight estimation accuracy.

The continuous monitoring of mean $S_v$ allows to follow properly density variations at least for the measured densities, up to the one third of the maximum possible in production cycles in the intensive Mediterranean aquaculture installations. Long time series during several months in a production cycle could help to establish the proper temporal window criteria for total biomass estimation and the density limits of its validity.

Fig 5: Estimated biomass density variations and constant-assumed density (horizontal line) from catches sequence for 541 mean weight sea breams in a “commercial” cage.

4. ACKNOWLEDGEMENTS

We acknowledge the fundamental collaboration of Grupo Culmarex, and the personnel of their aquaculture plants of Águilas (Murcia, Spain) and Piagua (Almería, Spain). Special thanks to Biosonics Inc. for providing the acoustical data token during their tests in Águilas as well as to the assistance of CTAQUA for the sea campaigns. This work was realised in the frame of the project ARM/1790/010.

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SOUND PROPAGATION OVER AN ELASTIC BOTTOM –
PARTICLE MOTIONS CAUSED BY SEISMIC INTERFACE WAVES

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Abstract: Particle motion sensitivity may be important for fish responding to low frequency anthropogenic such as sounds generated by piling and explosions. This article discusses particle motions of seismic interface waves generated by low frequency sources close to solid rigid bottoms. The interface waves are transversal waves with slow propagation speed and characterized with large particle movements, particularity in the vertical direction. The waves decay exponentially with distance from the bottom. The interface waves may be important to include in the discussion when studying the impact of low frequency anthropogenic noise at generated by relative low frequencies, for instance by piling and explosive charges, airguns and subsea construction works.

Keywords: interface waves, seismic noise, impact on marine life

1. INTRODUCTION

Particle motion sensitivity may be important for fish responding to low frequency anthropogenic such as sounds generated by piling and explosions [1] It is therefore surprising that studies of the impact of such sounds upon fish and invertebrates have usually focused on propagated sound pressure, rather than particle motion [2]. It appears that the importance of interface waves have not been recognized and discussed in detail in relation to impact on marine life, with the exception of the paper [3], which inspired this study. The purpose in this article is to discuss in more details, the excitation and propagation of transversal waves propagating along an interface between water and a solid medium.
2. THEORY ON SOUND PROPAGATION OVER A SOLID BOTTOM

Fig. 1 shows the situation under consideration. A point source is located a height $z_s$ above the sea floor and the receiving point is at a horizontal distance $r$ at a height $z$ above the bottom. The sea bottom may be composed of any number of layers and modeled as either fluid or solid layers with densities $\rho_n$, sound speeds $c_{pn}$ and shear speeds $c_{sn}$ where $n = 1, 2, \ldots, N$ for the layers in the bottom.

The received signals may have several components, a direct and a bottom reflected signal, both indicated with dashed lines in the figure. Then there is a refracted signal, indicated by a solid blue line, entering the bottom at critical angle and propagating along the interface. These waves exist also with a soft fluid-like bottom with no rigidity and no shear waves. With a solid bottom, defined as a bottom that can support shear waves, there will also be transversal waves propagating along the interfaces at speeds somewhat lower than the shear speeds of the seafloor.

The received signal $\phi(r,z,\omega)$ with angular frequency $\omega$ at depth $z$, range $r$ is expressed by an integral over the horizontal wave numbers [4],

$$
\phi(r,z,\omega) = \frac{S(\omega)}{4\pi i} \int_0^\infty \frac{\exp(i\gamma_0|z - z_s|)}{\gamma_0} kJ_0(kr)dk
+ \frac{S(\omega)}{8\pi i} \int_0^\infty \frac{R_b(k)}{\gamma_0} \exp(i\gamma_0|z + z_s|) kH_0^{(1)}(kr)dk
$$

$J_0(kr)$ is the 0-order Bessel function of first kind and $H_0^{(1)}(kr)$ is the 0-order Hankel function of the first kind representing an outgoing cylindrical wave. $S(\omega)$ is the frequency function of the source signal. The vertical wave number component in the water is $\gamma_0$, and the horizontal wave number component is $k$. These components are related to the frequency $\omega$ and the sound speed in the water $c_0$ by
\[ \gamma_0 = \sqrt{\left(\frac{\omega}{c_0}\right)^2 - k^2} \]  

The first term in equation (1) is the direct signal from the source, and the second term is the signal reflected from the medium below. The reflectivity of the seafloor, defined by the plane wave reflection coefficient \( R_b(k) \), is a function of the horizontal wave number component \( k \). When \( \phi(r,z,\omega) \) denotes the velocity potential, the normal stress, equal to negative pressure is

\[ \sigma = -\rho \frac{\partial}{\partial t} \phi(r,z,\omega) = i\omega\rho \phi(r,z,\omega) \]  

The vertical \( u_z \) and the horizontal particle velocities \( u_r \) are given by

\[ u_z = \frac{\partial}{\partial z} \phi(r,z,\omega) \]
\[ u_r = \frac{\partial}{\partial r} \phi(r,z,\omega) \]  

3. AN EXAMPLE WITH A POINT SOURCE OVER A SOLID HALF SPACE

Consider first the simple case with only one elastic layer having compressional wave speed \( c_{p1} \), shear speed \( c_{s1} \) and density \( \rho_1 \). The sound speed in the water is \( c_0 \) and the density is \( \rho_0 \). The second term in equation (1) contains at least two major contributions; when \( R_b(k) \) is very large and when the vertical wave number \( \gamma_0 \) is zero. The maximum values of \( R_b(k) \) can be found numerically and it turns out that the maximum value approximately occurs when

\[ k \approx \frac{\omega}{c_s} \]  

The corresponding phase velocity is therefore near equal to the shear wave speed of the upper strata of the bottom. A more detailed analysis shows that phase velocity is a little lower, about 95%, of the shear speed. Equation (5) implies that a slow-propagating wave can exist at the interface between the water and the solid bottom having a speed slightly lower than the shear speed of the bottom material. In addition to these two wave components, there is also a refracted wave, but in the cases considered in this study the refracted arrivals are weak and can be ignored, but can actually be observed in some of the simulated pulse plots shown later. Inserting equation (5) into equation (2) shows that the vertical wave number in the water \( \gamma_0 \) is imaginary. When \( c_{s1} < c_0 \), as often is the case, the amplitude of the interface wave decays exponentially with the distance from the interface according to,

\[ a(z) \propto \exp \left( -\frac{\omega}{c_{s1}} |z + z_i| \right) \]  

Fig. 2: shows a plot of integrand of second term of equation (1). The red dotted line marks the location of the poles of the interface wave; the green and blue dotted line marks the reflected and the refracted arrivals. For instance, at the frequency of 100 Hz the three wave components have peaks for the values of \( k \) equal to 1.59, 0.42 and 0.315
corresponding to phase velocities 395 m/s, 1500 m/s and 2000 m/s for the interface wave, the reflected wave, and the refracted waves, respectively. Notice that the green line, the line for the sound speed in the water, gives the limit of real incident angles. This means that interface waves are not excited by plane wave incident, but by spherical waves as a near field effect. The fact that the peaks are on straight lines means that the waves propagate with constant velocities independent of frequency and therefore the waves are not dispersive.

![Graph showing phase velocities](image)

**Fig. 2:** Absolute value of the integrand for a homogeneous elastic bottom as function of the horizontal wave number and frequency. The straight lines indicate the contribution of the interface waves and the reflected and refracted waves as given in the legend.

### 4. CASE STUDIES USING THE OASES MODEL

Numerical solutions are obtained using the wavenumber integration technique [5], which is implemented in the OASES model [6, 7]. This model is now used to illustrate how the sound pressure and the particle velocities may vary with the properties of the bottom. The source signal is a 5 Hz Ricker pulse, released at a height of 5 m above the bottom, which is at 100 m depth. In all cases, the receiver is at 99 m, 1 m above the bottom.

Consider first the case with a fluid homogeneous bottom with density $\rho_1=2000 \text{ kg/m}^3$ and sound speed $c_{p1} = 2000 \text{ m/s}$. Fig. 3 shows the received pulses at points 1 m above the bottom at different ranges of the normal stress component and the vertical and horizontal particle velocities. The very first, almost invisible, arrivals are the refracted waves arriving just before the strong direct and bottom reflected pulses. In these plots, compensation for spherical spreading is applied by proportionally increasing the amplitudes with range. Therefore, the amplitudes of the direct and reflected pulse shapes appear to be constant with range.

The effect of shear waves are illustrated in Fig. 5: by considering the case of a solid bottom with shear speed $c_{s1} = 400 \text{ m/s}$ and attenuations of 0.2 dB /wavelength. The three components of the received signals all exhibit strong low-speed interface contributions, in particular for the vertical particle velocity component.

The third case, displayed in Fig. 5:, is with a layered bottom with 20 m layer over a solid half-space. The parameters of the upper layer are the same as used to generate Fig. 4:, but the solid half-space has a density of $\rho_2=2200 \text{ kg/m}^3$, compressional wave speed $c_{p2} = 2200 \text{ m/s}$, and shear speed $c_{s2} = 600 \text{ m/s}$, the attenuations of both the compressional and the shear waves in the bottom are 0.2 dB per wavelength. The interface waves are in this
case dispersive caused by the depth dependence of the shear speed from 400 m/s at the surface of the bottom to 600 m/s at 20 m depth into the bottom.

Fig. 3: Pulse responses over a fluid bottom at depth of 100m. Normal stress, vertical particle velocity, and horizontal particle velocity as function of range from the source.

Fig. 4: Pulse responses over a homogenous elastic bottom at depth of 100 m. Normal stress, vertical particle velocity, and horizontal particle velocity as function of range from the source.

Fig. 5: Pulse responses over a layered elastic bottom at depth of 100 m. Normal stress, vertical particle velocity, and horizontal particle velocity as function of range from the source.
5. SUMMARY AND CONCLUSIONS

This article has discussed particle motions of seismic interface waves generated by low frequency sources close to solid rigid bottoms. Seismic interface waves are transversal waves in the sagittal plane with significant particle motions, particularly in the vertical component. The strength and propagation of interface waves are dependent on the seismo-acoustic properties of the bottom, particularly the shear speed and attenuations of the bottom. The amplitude of the interface waves decays exponentially with distance from the bottom and may therefore have effect on marine life on the bottom or very close to the bottom. Particle motion sensitivity may be important for fish responding to low frequency anthropogenic such as sounds generated by piling, airguns and explosive charges. At present, very little work has been carried out on the sensitivity of fish and other organisms, including marine invertebrates, to particle motion. It is therefore recommended that future experiments be conducted with three-axis vector sensors in addition to hydrophones to monitor the full acoustic field.

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Underwater Communication
DATA SMOOTHING ALGORITHMS FOR
PHASED ARRAY DOPPLER LOG

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Abstract: Doppler log is one of the navigation equipments on ships. With the development of marine technology, more and more countries and institutions began to study it, especially the one for deep sea environment. Velocity compensation was once a bottleneck to improve the accuracy. The application of phased-array technique on sonar solved this problem in theory, which made the performance of Doppler log greatly improved. The precision of speed which is measured by Doppler log is high, but in some situations there are some outliers of the measured doppler value. In order to obtain the high accuracy speed result, some smooth processing algorithms should be applied to the speed estimated. The commonly smooth method is interchange method, average value filtering and so on, but these methods are only suitable for that wild value spots are few. Three methods for smooth filtering that moved window and weighted mean method, circle sieve method, kalman filtering are studied in this paper. The relation of outliers’ elimination and tracing speed are considered. The situation of the lake test is introduced, and the performances and application conditions of the algorithms are analyzed based on the experimentation condition. Finally, Kalman filter is realized on the real-time processor and work well.

Keyword: Phased Array Doppler Log, Data Smoothing algorithms, circle sieve, kalman, moving window weighted average
1 INTRODUCTION

The application of phased-array technique on sonar solved velocity compensation problem in theory, which made the performance of phased array doppler log greatly improved. Along with the Doppler velocity log widely applying, the precision of measured velocity has also become the hot spot which people pay attention. The precision of speed which is measured by Doppler log is high. Doppler log can also obtain speed of the opposite seabed\(^1\), \(^2\). But in some situations there are some outer data (wild value spot) in the Doppler velocity log. The so-called outer data which has different characteristics with the majority data is individual data in the data. The outer data possibly deviate from carrier speed, random error statistical property of the outer data is possibly different from the majority data. In order to obtain the high accuracy speed, this needs to do smooth processing to the Doppler navigation data. Reject the outer data and obtains the smooth curve.

Generally, the commonly used smooth method is interchange method, average value filtering and so on, but these methods are only suitable for that wild value spots are few. This article proposed three methods for smooth filtering that moved window and weighted mean method, circle sieve method, kalman filtering. Three methods can smooth data. The formal two methods cannot process the first several spots in the data, but the Kalman filtering can solve that problem. The Kalman filtering method is realized on TMS320C6455 development board. The Kalman filtering has the very good timeliness.

2 DATA SMOOTHING ALGORITHMS

There are many uncertain factors in the lake trial, such as waves, fish, and array jitter, leading to error speed measuring results. For a ship, its speed is in a certain range, the acceleration also has a limit, therefore when the estimation result is beyond the boundary of a certain value, it can be regarded as outliers, besides, if the data is in the range of the boundary, the velocity estimated is either right value or error value.

On one hand, for the outliers beyond the range of ship velocity, it can be eliminated directly, and we can replace it with the value calculated by several right results. On the other hand, for the data within the scope of the ship velocity, we can adopt some smoothing algorithms which can only get some processing gain but also Inhibition of outliers. Three smoothing algorithms will be introduced in the following.

2.1 CIRCLE SIEVE METHOD

The idea of "circle sieve " picking wild points comes from the impulse denoise in image processing, which requires effective samples is more than 50%. Assume \( f(n) \) is the sample data, in which \( 1 < n < N \), \( N \) is total number of samples to be processed. \( f(n_k) \) is the k indexed data, in which \( N_0 \leq k \leq N \). In order to distinguish it is wild data or not, we should use data \( f(n_i) \), in which \( k-N_0 < i < k-1 \). \( Q \) and \( Q_0 \) are the criterion equation shown as
below.

\[
Q = \left( \sum_{i=1}^{N_k} (n_k - n_{k-i})^2 + (f(n_k) - f(n_{k-i}))^2 \right), \quad N_0 \leq k \leq N
\]  

(1)

\[
Q_0 = \left( \sum_{i=1}^{N_k} (n_k - n_{k-i})^2 \right) + N_0 \ast b^2, \quad N_0 \leq k \leq N
\]  

(2)

Where \( b \) is tolerance limit in equation (2).

\[
\mu = \frac{1}{Q}, \quad \mu_0 = \frac{1}{Q_0}
\]  

(3)

Where \( \mu_0 \) is threshold of samples. If \( \mu \geq \mu_0 \), the data is effective, or it is considered to be wild value. Usually in practical application, \( \mu_0 \) will be multiple with a coefficient \( \varepsilon \) which is computed by portion of effective data in total samples, so \( 0 < \varepsilon \leq 1 \), and the more effective data, the value of \( \varepsilon \) incline to 1. Assume \( l(n_k) \) is the results after circle sieve process. So we get equation (4).

\[
\begin{cases}
  l(n_k) = f(n_k), \quad \mu \geq \mu_0 \ast \varepsilon \\
  l(n_k) = f(n_{k-1}), \quad \mu \leq \mu_0 \ast \varepsilon
\end{cases}
\]  

(4)

2.2 MOVING WINDOW WEIGHTED AVERAGE METHOD

The velocity \( v_k (k = 1, 2, 3 \ldots N) \) measured by Doppler log is composed of real ship velocity and random error. We consider the error as Gaussian noise. \( v_k \) is divided into some frames, and each frame contains \( n \) data as notated \( v_{m-n+1}, v_{m-n+2}, v_{m-n+3}, \ldots, v_m \) the median value of each frame is give in equation (5).

\[
median(v_m) = \begin{cases}
  \frac{v_m(n/2) + v_m(n/2+1)}{2}, \quad \text{n is even number} \\
  v_m(n/2), \quad \text{n is odd number}
\end{cases}
\]  

(5)

\[
r_m = v_m - median(v_m)
\]  

(6)

In equation (6) the residual \( r_m \) is the difference of output value and median value. Because the random error is Gaussian, so the residual \( r_m \) is also obeyed by Gaussian distribution. There is a Empirical coefficient 1.3 between The standard deviation \( \sigma_m \) and mean of residual value \( r_m \), so we get equation (7).

\[
\sigma_m = 1.3 \ast mean(|r_m|) + 10^{-12}
\]  

(7)

The Constant coefficient \( 10^{-12} \) in (7) is to avoid get \( \sigma_m \) value equals to 0. We define a threshold \( th_m \) by \( \sigma_m \) as shown in equation (8).

\[
th_m = 4.6 \ast \sigma_m
\]  

(8)
where \( w_m \) is a weighted coefficient. The less of value \( |r_m(i)| \), the more \( w_m \) inclines to 1. Equation (10) is the final value after moving window weighted average method.

\[
z_m(j) = \frac{\sum_{i=m-n+1}^{m} v_m(i) \cdot w_m(i)}{\sum_{i=m-n+1}^{m} w_m(i)}
\]

### 2.3 KALMAN METHOD

The Kalman filter is a set of mathematical equations that provides an efficient recursive means to estimate the state of a process, in a way that minimizes the mean of the squared error (MMSE). It estimates a process by using a form of feedback control: the filter estimates the process state at some time and then obtains feedback in the form of noisy measurements. Kalman filter is widely used in many aspects of real-time signal processing because the value of system state estimation can be updated realtime.

Kalman filter equations include two groups: time update equations and measurement update equations. The time update equations are responsible for projecting forward the current state in time and error covariance estimates to obtain the a priori estimates for the next time step. The measurement update equations are responsible for the feedback—i.e. for incorporating a new measurement into the a priori estimate to obtain an improved a posteriori estimate. The time update equations can also be thought of as predictor equations, while the measurement update equations can be thought of as corrector equations. Indeed the final estimation algorithm resembles that of a predictor-corrector algorithm for solving numerical problems. A complete picture of the operation of the filter is shown in Fig.1[3].
3 SIMULATION RESULTS

In order to test the performance of data smoothing algorithms for phased array Doppler log, we made an experiment in Qiandao lake, Zhejiang province, China. The lake is about sixty meters depth. We adopt three smoothing methods to the velocity estimated by Doppler log. The processing flow chart is shown in Fig.2. First the original velocity data is processed by circle sieve method to exclude the wild value data, then kalman and moving window weighted average algorithm is adopted to smooth the data. The comparison result with GPS is also given.

![Flow chart of smoothing algorithms](image)

**Fig.2** The smoothing algorithms processing flow chart

3.1 CIRCLE SIEVE METHOD SIMULATION

Fig.3 shows the results after processing by circle sieve method, from the red line we can see that most wild value are excluded. And the wild value is replaced by last right value. The parameters of circle sieve is that \( N_0 = 3, \  \epsilon = 0.8 \).

![Circle sieve method result](image)

**Fig.3** wild value excluded by circle sieve method

3.2 SMOOTHING ALGORITHMS

From Fig.3 we can see that most wild value are excluded, but the data is still affected by noise and further processing should be used to get better result. We adopt kalman and moving window weighted average to smooth the data. The results are shown in Fig.4 and Fig.5. Besides, we compare the smooth results with the velocity measured by GPS which is considered as standard value. Fig.4 and Fig.5 also show the error by minus the processing curve with GPS curve. The error is less than 0.2 knot.
4 CONCLUSION

Though phased array Doppler has high precision in velocity measured, there are various noise source in the water, complex bottom or current in the sea and ship sway induced by wind and waves. Its original velocity results computed from the received signal has high error. Some smoothing algorithms should be used in the system to get better results. In this paper, three algorithms are proposed to process the velocity measured in the lake, less than 0.2 knot error was achieved.

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REFERENCE

COMPARISON OF MODULATION TECHNIQUES FOR PARAMETRIC UNDERWATER COMMUNICATIONS

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Abstract: In parametric underwater communications, nonlinear effects occurring during the wave propagation in the underwater acoustic channel are particular used for data transmission. One point of interest in parametric underwater communications is that different modulation techniques can be applied to generate identical received signals, which is inherently due to the underlying nonlinear channel. This paper shows that different modulation techniques which create identical received signals achieve different signal conversion efficiencies. Consequently, different transmit powers are required to excite the nonlinear signal generation to the same amount. One outcome is that for quadrature-amplitude-modulation the transmission of two modulated signals using correlated data sequences outperforms the conventional approach, which is the transmission of one modulated signal and an additional carrier. The results are verified by measurements, which are conducted using a model parametric communication system in air.

Keywords: parametric underwater communications, communication over nonlinear channels, parametric transduction

1. INTRODUCTION

Parametric underwater communications (PUC) is an alternative approach for realising digital data transmission in underwater scenarios. The key idea is the particular usage of nonlinear effects which occur during the wave propagation in the underwater acoustic channel. For this, an appropriately modulated, high-frequency wave is radiated by a transducer. Due to intermodulation in the wave field, new frequency components are generated during the wave propagation. Among others, low frequency components containing the transmitted information are generated which can propagate over long
distances. Furthermore, corresponding systems benefit from a high relative bandwidth and a high directivity while using a small transducer aperture.

From the communication point of view, the PUC bases on a nonlinear channel. Two major requirements that have to be considered in the system design arise from the nonlinearity: Firstly, the nonlinear signal distortions eventually induce an information loss. For this reason, appropriate modulation techniques have to be applied. Secondly, the nonlinear signal conversion process creates a signal with only a very low signal power. Moreover, only low frequency components of the nonlinearly generated signal can be used for communication purposes in the underwater scenario, which further decreases the power efficiency of PUC systems. Consequently, signal processing techniques to enhance the power efficiency of PUC are of particular interests.

Outstanding in the PUC is that different transmitted signals can create identical signals at the receiver which is due to the inherent nonlinear channel [1]. Moreover, it will be shown in this paper that the different modulation techniques achieve different signal conversion efficiencies. In this context, the term signal conversion efficiency describes which quantity of the transmit power is converted to the nonlinearly generated signal. Since the conversion efficiency is a critical aspect in the PUC and no corresponding analysis of modulation techniques can be found in the literature, modulation techniques are compared in terms of their signal conversion efficiency in this paper. The focus is on linear modulation techniques that utilise the nonlinear distortion for the signal generation.

The discussion in this paper is restricted to the comparison of two basic modulation techniques which were commonly used in PUC systems [1]. For the comparison, the transmit powers that are required by the different modulation techniques to generate a certain signal after the nonlinear distortion are calculated and compared with each other.

It will be shown that for amplitude and quadrature-amplitude modulation (QAM), the transmission of two modulated signals using correlated data sequences outperforms the conventional approach, which is the transmission of one modulated signal and an additional carrier. Contrarily, using phase modulation in combination with rectangular pulse shaping, the same conversion efficiency will be obtained.

The paper is structured as follows: In Section 2, a channel model of the parametric transduction process and the considered basic modulation techniques are briefly introduced. Subsequently, the modulation techniques are generalized to one general approach, which is discussed in Section 3. Analytical and measurement results are discussed in Section 4 and Section 5 as well. The paper concludes in Section 6.

2. COMMUNICATION OVER THE NONLINEAR PARAMETRIC CHANNEL

The PUC bases on the parametric transduction [2]. The nonlinear transduction principle is used in several application fields, for instance sub button profiling. In [3], a point-to-point model for the underlying parametric channel is derived from a physical modelling using a grey-box modelling approach. The proposed model consists of the concatenation of a static second-order nonlinearity and a dynamic linearity having a highpass transfer characteristic, see Figure 1. Although this general channel model gives only qualitative insights in the parametric transduction process and no further characteristics of the acoustic underwater channel are considered, it is sufficient to understand the basic idea of the linear modulation techniques discussed in the following.

The first modulation technique is denoted by conventional approach. This approach has been realised and tested in practical PUC systems realizations [4]. Its principle bases on the transmission of a linearly modulated signal at the centre frequency $f_1$ superimposed
by an unmodulated carrier at the centre frequency $f_2$. With $d$ denoting an arbitrary data sequence to be transmitted and $h(t)$ being the impulse response of the transmit filter, the corresponding transmitted signal $x^{\text{ca}}(t)$ may be written in the form

$$
\frac{1}{\sqrt{2}} x^{\text{ca}}(t) = \Re \left\{ \sum_{k=-\infty}^{\infty} d[k] h(t - kT_s) e^{j2\pi f_1 t} \right\} + \Re \left\{ e^{j2\pi f_2 t} \right\} = \Re \left\{ s(t) e^{j2\pi f_1 t} \right\} + \Re \left\{ e^{j2\pi f_2 t} \right\},
$$

(1)

where $s(t)$ is the equivalent lowpass signal created by an arbitrary linear modulation and $T_s$ denotes the symbol duration. According to the channel model in Figure 1, the nonlinear distorted signal $y(t)$ reads after some manipulations

$$
y^{\text{ca}}(t) = 1 + \frac{1}{2} |s(t)|^2 + \Re \left\{ s(t) e^{j2\pi f_1 t} \right\} + \Re \left\{ s(t) e^{j2\pi f_1 t + f_2 T_s} + s(t) e^{j4\pi f_1 t} + e^{j4\pi f_2 t} \right\}.
$$

(2)

The first two terms of the sum in the right hand side (RHS) of equation (2) are low frequency signals and will be significantly attenuated by the subsequent highpass filtering. Furthermore, the signal components described by the fourth term in the RHS of equation (2) are high frequency signals. Considering long range transmission over the underwater acoustic channel, these components feature a vanishingly low power due to the frequency dependent channel attenuation. Contrarily, the signal component described by the third term on the RHS of equation (2) is a middle frequency signal and can propagate over comparative long distances. Neglecting all further channel characteristics, a received signal $r(t)$ of the form

$$
r(t) \approx \Re \left\{ s(t) e^{j2\pi f_1 t} \right\} = \Re \left\{ \sum_{k=-\infty}^{\infty} d[k] h(t - kT_s) e^{j2\pi f_1 t} \right\},
$$

(3)

will be obtained after the parametric transmission over long distances. It can be seen that the equivalent lowpass signal $s(t)$ is contained in the received signal and can be demodulated by bandpass-lowpass transform with the reference frequency $|f_1 - f_2|$. This way, a data sequence with an arbitrary constellation can be transmitted over the nonlinear channel without information loss.

In another approach two linearly modulated signals at the centre frequencies $f_1$ and $f_2$, respectively, are transmitted. The transmitted signal $x^{\text{pa}}(t)$ may be written in the form

$$
\frac{1}{\sqrt{2}} x^{\text{pa}}(t) = \Re \left\{ \sqrt{2} |s(t)| e^{j2\pi f_1 t} \right\} + \Re \left\{ e^{j2\pi f_2 t} \right\}.
$$

(4)

It can be seen that the signals are correlated with each other and that the equivalent lowpass signal $s(t)$, which shall be contained in the received signal, occurs predistorted in the transmitted signal. Thus, this technique is denoted by modulation by predistortion. Accordingly, the nonlinear distorted transmitted signal reads after some manipulations

$$
y^{\text{pa}}(t) = |s(t)| + \Re \left\{ s(t) e^{j2\pi f_1 t} \right\} + \Re \left\{ s(t) e^{j2\pi f_1 t + f_2 T_s} + s(t) e^{j4\pi f_1 t} + s(t) e^{j4\pi f_2 t} \right\}.
$$

(5)

Again neglecting signal components at centre frequency zero and high frequency components, i.e., the first and the third term of the sum in the RHS of equation (5), a received signal equivalent to equation (3) is obtained.

In the following investigation, constellation scrambling is applied at the transmitter for the modulation by predistortion approach. For further reading the reader is referred to [1].
3. GENERALIZATION OF LINEAR MODULATION TECHNIQUES

The introduced modulation techniques can be derived from the general approach

\[
\frac{1}{T_s} x(t) = \text{Re}\{s_1(t)e^{j2\pi f_1 t}\} + \text{Re}\{s_2(t)e^{j2\pi f_2 t}\}
\]

\[
= \text{Re}\left\{s_1(t)e^{j2\pi f_1 t} + s_2(t)e^{j2\pi f_2 t}\right\} e^{j2\pi f_c t}, \quad f_d = |f_1 - f_2|, \quad f_c = \frac{1}{2} (f_1 + f_2),
\]

(6)

where \(s_1(t)\) and \(s_2(t)\) are equivalent lowpass signals for the reference frequencies \(f_1\) and \(f_2\), respectively. Since \(s(t)\) is linearly modulated in equation (1) and (4), the equivalent lowpass signals \(s_1(t)\) and \(s_2(t)\) in equation (6) are generally of the form

\[
s_v(t) = \sum_{k=-\infty}^{\infty} d_v[k] g_v(t - kT_s), \quad v \in \{1, 2\},
\]

(7)

with \(d_v\) denoting two arbitrary data sequences and \(g_v(t)\) being the impulse responses of two transmit filters. The particular modulation techniques can be derived from equation (7) by parameterising the data sequences \(d_v\) and the impulse responses \(g_v(t)\) appropriately.

To calculate the power density spectrum of an arbitrary signal generated by the general approach in equation (6), the autocorrelation function \(R_{xx}(t + \tau, t) = E\{s^*(t) s(t + \tau)\}\) is determined at first, giving a sum of four terms. With \(m, n \in \{1, 2\}\), an arbitrary term of the sum may be written in the form

\[
\sum_{\lambda=-\infty}^{\infty} R_{ddmnu}(\lambda) \sum_{k=-\infty}^{\infty} g_m^*(t - kT_s) g_n^*(t + \tau - kT_s - \lambda T_s) e^{j(n - m)2\pi f_0 \delta t} e^{j(2n - 3)\pi \lambda \tau}.
\]

(8)

The inner sum in equation (8) is periodic in time with the period \(T_s\) and hence the autocorrelation function is periodic in time with period \(T_s\). To calculate the power density spectrum of the cyclostationary process, equation (8) is averaged over one period, reading

\[
\frac{1}{T_s} e^{j(2n - m)\pi f_0 \delta t} \sum_{\lambda=-\infty}^{\infty} R_{ddmnu}(\lambda) \int_{t=-\infty}^{\infty} g_m^*(t) e^{j(n - m)2\pi f_0 \delta t} g_n^*(t + \tau - \lambda T_s) \, dt.
\]

(9)

It is assumed in equation (9) that the ratio \(T_s/T_0\) is integer, which simplifies the derivation significantly. Next, equation (9) is transformed into the frequency domain, reading

\[
\frac{1}{T_s} G_m^*(f - (2m - 3)\delta f) G_n(f - (2n - 3)\delta f) \sum_{\lambda=-\infty}^{\infty} R_{ddmnu}(\lambda) e^{-j2\pi f \lambda T_s}.
\]

(10)

The summation of the terms in equation (10) for \(m, n \in \{1, 2\}\) leads after some manipulations to the power density spectrum \(S_{xx}(f)\) of the equivalent lowpass signal \(s(t)\)

\[
S_{xx}(f) = \sum_{\lambda=-\infty}^{\infty} r_{dd1}\left(\lambda\right) e^{-j2\pi f \lambda T_s} + \sum_{\lambda=-\infty}^{\infty} r_{dd2}\left(\lambda\right) e^{-j2\pi f \lambda T_s}.
\]

(11)

and the power density spectrum of the transmitted signal \(x(t)\) can be subsequently determined from \(S_{xx}(f) = \frac{1}{T_s} S_{xx}\left(f - f_c\right) + \frac{1}{T_s} S_{xx}\left(f + f_c\right)\). Since the lowpass-bandpass transform does not change signal power, equation (11) can be directly used to determine the transmit power by applying Parseval’s theorem.
4. COMPARISON OF MODULATION EFFICIENCY

An elementary modulation scheme is considered to illustrate the approach for the comparison of the signal conversion efficiencies. For this, a linearly modulated signal featuring an on-off-keying (OOK) constellation and rectangular pulses shall be generated after the nonlinear distortion. This means, the equivalent lowpass signal in the received signal \( s(t) \) features \( \{0, \sqrt{2}\} \) and \( h(t) = \text{rect}(t/T_s) \).

To employ the general approach of (6) corresponding to the conventional approach, the first transmitter equivalent lowpass signal is \( s_1(t) = s(t) \). This requires \( d_1 = d \in \{0, \sqrt{2}\} \) and \( g_1(t) = h(t) = \text{rect}(t/T_s) \). In order to generate an unmodulated carrier at frequency \( f_2 \) after the lowpass-bandpass transform, \( s_2(t) \) has to be equal to one, which can be achieved by defining \( d_2 = 1 \) and \( g_2(t) = \text{rect}(t/T_s) \). Substituting the resulting correlation functions, i.e., \( r_{\text{d}1}(\lambda) = \frac{1}{2} \delta(\lambda) + \frac{1}{2} \), \( r_{\text{d}2}(\lambda) = 1 \) and \( r_{\text{d}12}(\lambda) = 1/\sqrt{2} \), as well as the transfer functions \( G_s(f) = F\{g_s(t)\} = T_s \sin(\pi f T_s) \) into equation (11), the power density spectrum

\[
S_{d2}^{\text{ca}}(f) = \frac{1}{T_s} \left[ \pi T_s \right] \sum_{\lambda=-\infty}^{\infty} \left( \frac{1}{2} \right) e^{j2\pi f T_s} + \frac{1}{T_s} \left[ \pi T_s \right] \sum_{\lambda=-\infty}^{\infty} e^{j2\pi f T_s} \tag{12}
\]

for the conventional approach is obtained. The third term in equation (11) contains the product of two si-functions, which is zero at multiples of \( 1/T_s \). Since it is multiplied by a Dirac-comb with period \( 1/T_s \), the cross correlation term in equation (11) reduces to zero. The integration of equation (12) determines the signal power to be \( P_{\text{ca}} = 1 \).

To employ the general approach of (6) corresponding to the modulation by predistortion, the equivalent lowpass signals have to be \( s_1(t) = s_2(t) = \sqrt{2}(t) \), which means that \( d_1 = \sqrt{2} \in \{0, \sqrt{2}\} \) and \( g_1(t) = \sqrt{2}h(t) = \text{rect}(t/T_s) \). This leads to the correlation functions \( r_{\text{d}1}(\lambda) = r_{\text{d}2}(\lambda) = r_{\text{d}12}(\lambda) = \frac{1}{\sqrt{2}} \delta(\lambda) \), giving the power density spectrum

\[
S_{d2}^{\text{pa}}(f) = \frac{1}{\sqrt{2} T_s} \left[ \pi T_s \right] \left( \frac{f + f_d}{2} \right) + \frac{1}{T_s} \left[ \pi T_s \right] \left( \frac{f - f_d}{2} \right) \tag{13}
\]

where again the cross correlation term is zero. The integration of equation (13) determines the signal power to be \( P_{\text{pa}} = \sqrt{2} \).

It can be seen from the ratio of the required transmit powers \( \Pi_{T_s,\text{OOK}}^{\text{ca,pa}} = P_{\text{ca}}/P_{\text{pa}} = \sqrt{2} \) that the transmit power required by the conventional approach is \( \sqrt{2} \)-times the transmit power required by the modulation by predistortion technique. This is a major outcome of the comparison which will be compared with measurement results in the following.

In the measurement two signals are generated, each corresponding to one modulation technique introduced in Section 2, to transmit randomly generated OOK-symbols with rectangular pulses over the nonlinear parametric channel. The correlation coefficient of the transmitted signals was determined to be \( \rho = 0.86 \), underlining the fact that actually two different signals are transmitted. For the parametric transmission, the air parametric transduction system reported in [3] is employed. Table 1 shows the used parameterization. At the receiver a bandpass-lowpass transform with reference frequency \( f_d = 1\text{kHz} \) is processed followed by a matched-filtering. The signal transmission is ten times repeated, enabling noise reduction at the receiver by averaging.

The correlation coefficient of the averaged received signals was \( \rho = 0.996 \), which shows that identical received signals were generated. The missing one per cent can be well explained due to noise. The power ratio of the averaged received signals was \( \Pi_{T_s,\text{OOK}}^{\text{ca,pa}} = 1.002 \). Consequently, the modulation by predistortion technique significantly outperformed the conventional approach for the applied OOK-modulation scheme.
5. RESULTS FOR QUADRATURE-AMPLITUDE-MODULATION

Now, a signal featuring a QAM constellation with rectangular pulses shall be generated after the nonlinear distortion. The approach is analogous to the one discussed in Section 4 and thus only the results are stated here. The ratios of the required transmit powers for \( M \)-QAM of commonly used constellation sizes are shown in Table 2.

It was shown in [1] that for \( M \)-QAM schemes the performance of the modulation by predistortion technique increases with increasing constellation sizes. Interestingly, the same values as shown in Table 2 were obtained. This indicates a general dependence between the signal conversion efficiency and the degree of amplitude modulation. Consequently, if a \( M \)-QAM scheme is applied, modulation by predistortion outperforms the conventional approach since the latter has a lower degree of amplitude modulation due to the unmodulated carrier.

This agrees with the fact that for \( M \)-ary phase-shift-keying (\( M \)-PSK) modulation, the ratio of required transmit powers is \( \Pi_{\text{Tx,PSK}}^{\text{ca,pa}} = 1 \), which means that the two considered modulation techniques show no difference in their modulation efficiency. That is because the transmitted signals possess no amplitude modulation at all.

\[
\begin{array}{cccc}
M & 4 & 16 & 64 \\
\Pi_{\text{Tx, } M\text{-QAM}}^{\text{ca,pa}} & 1 & 1.056 & 1.065 & 1.067 \\
\end{array}
\]

Table 2: Ratio of required transmit powers for generating \( M \)-QAM schemes.

6. CONCLUSION

It was shown in this paper that the degree of amplitude modulation influences the signal conversion efficiency. As a result, the modulation by predistortion outperforms the conventional approach for \( M \)-QAM schemes. This is not the case for \( M \)-PSK modulation schemes, where no difference is achieved.

The proposed approach can be applied to derive and compare further appropriate modulation techniques for PUC.

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INQUIRING FLOODING ALGORITHM FOR
UNDERWATER ACOUSTIC SENSOR SELF-ORGANIZATION NETWORK

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Abstract: Usually, UASN (Underwater Acoustic Sensor Network) is constructed by randomly deployed sensor nodes. In order to construct network of certain function with these nodes, self-organization algorithm is needed. There are many self-organization algorithms for WSN (Wireless Sensor Network) onshore. But, in underwater acoustic communication, the phenomena, such as: severe attenuation, high background noise, limited bandwidth, large time delay, complicated multi-path, etc., make UASN different from WSN. Most self-organized algorithms for WSN are hard to be applicable to UASN. For example, in OPNET simulation of flooding self-organization algorithm in UASN, it takes an unreasonable long time to establish an effectual network, which means each node could be accessed by others. Even worse, the network can not be established without reference to the broadcast times of each node. An improved self-organization algorithm is proposed, which appends an inquiry process to flooding self-organization algorithm. It is proved by OPNET simulation that under the same conditions an effectual network could be successfully established in a shorter time than simple flooding and probabilistic flooding algorithms, and the energy consumed in self-organization period is decreased.

Keywords: Underwater Acoustic Sensor Network, Self-Organization Algorithm, Flooding
1. GENERAL

In common, UASN consists of several drifting or submersible buoys (treated as nodes in network) which are deployed by ship or plane. After deployment, self-organization algorithm is needed to construct certain UASN. There are two major ideas of wireless network self-organization. 1) Each buoy discovers routes to the other buoys, generates a route table and maintain it. 2) None of the buoys constructs route table. A proper route is established only when a packet needs to be transmitted. Considering the communication resource is restricted, the idea 1) is chosen.

The most popular scheme of self-organization in wireless sensor network is flooding. Flooding scheme is divided to four types in general[1]: 1) simple or blind flooding, 2) probabilistic or gossip flooding, 3) area based flooding, 4) neighbour knowledge flooding. Simple or blind flooding causes broadcast storm, which leads to channel congestion easily[2]. When area based flooding or neighbour knowledge flooding is used, it’s hard for submersible buoy to obtain geography or topology information after deployment[3][4].

Simple flooding (SF) and probabilistic flooding (PF) are both transplanted to UASN and simulated in OPNET. In most cases, part of the nodes can not construct an effectual route table within the specified time. Effectual route table means any other node could be accessed according to the route table. An inquiring flooding (IF) algorithm is proposed. Node inquires its neighbours of the route to interested node after broadcast “I’m alive” once. During simulation of several typical topologies, all nodes succeed in constructing an effectual route table within the specified time using IF algorithm.

2. ALGORITHM DESIGN

![Flow chart of inquiring flooding](image)

First of all, the important assumption of IF is the underwater acoustic communication channel is reciprocal. The flow chart of IF is shown in Fig.1, in which “RxData” stands for “received data” and “Effectual” for “route table is effectual, all the other nodes can be accessed”. By the word “INQUIRE”, we means that the node, which inquires the route to one of the inaccessible nodes, broadcasts an inquiry frame to its neighbours within one hop. The neighbour who knows the route and is idle meanwhile will reply the inquiry frame.
According to Fig.1, each node sets a random flooding interrupt during “initiation”, and waits for the interrupts in “INIT” state. When the interrupt occurs and route table is not effectual, flooding is launched. After flooding or the route table has been effectual, the node idles. According to the data received during “IDLE” state, route table will be updated and checked. If it is not effectual, neighbour nodes will be inquired.

![Fig.2: Flow chart of data processing](image)

The flow chart of data processing in receiving end is shown in Fig.2, in which “Inquiry?” and “Flooding?” are data type judgement. Bit error detection is required before data type judgement. If any bit error exists, the data is discarded without any process anymore. Otherwise, process the data according to Fig.2. Therefore, stop and wait transmission is recommended in transmitting end.

3. SIMULATION SETUP

Simulations were carried out in OPNET. Full-duplex wireless communication is modelled by 14 pipeline stages in OPNET. But, it isn’t applicable to underwater acoustic communication. Changes have to be made as follows.

- Propagation velocity must be changed to 1.5km/s in the 6th stage.
- Marsh-Schulkin model is adopted to calculate the propagation loss in the 8th stage.
- In the 9th stage, interference noise is amplified, in order to simulate the error bits caused by collision.
- In the 10th stage, background noise is computed according to the classical formula.
- In the 11th stage, the realistic bit error rate (BER) is supposed be $10^{-3}$ when no collision happened and $10^{-1}$ when collision happened. To achieve this, transmit power should be adjusted to a proper value.
- The half-duplex communication node model is designed as shown in Fig.3, in which “Source” stands for data source, “Route” for self-organization algorithm module. The direction of data packet is shown by solid lines. One of dashed lines is the rising edge statistic interrupt of data receiving; the other is the falling edge statistic interrupt. Half-duplex communication is realized by their cooperation.
Other parameters of the node model are set as follows: centre frequency 12kHz, bandwidth 4kHz, transmitting bit rate 1kbps, communication distance 5.5km, self-organization time limitation 1800s.

Four typical topologies in Fig.4 are set in the simulation. Self-organization time and energy consumption of simple flooding, probabilistic flooding and inquiring flooding are investigated. The probability of broadcast in probabilistic flooding is set to 0.5. The self-organization timer ends when the route table is checked to be effectual. Energy consumption contains the electric energy consumed in transmitting, receiving and standing by.

![Fig.4(a) Star topology](image)
![Fig.4(b) Ring topology](image)
![Fig.4(c) Cellular topology](image)
![Fig.4(d) Distributed topology](image)

**Fig.4 Topologies in the simulation**

4. RESULT ANALYSIS

The simulation results of four topologies and three algorithms are listed in Table.1. The “Broadcast times” and “Inquire times” columns indicate the times of broadcasting and inquiring respectively, while “Completion” means the number of nodes which has constructed an effectual route table within the time limitation 1800s.

For all of the four topologies, none of the network adopting simple or probabilistic flooding algorithm has constructed effectual route table within the time limitation. For the network adopting inquiring flooding algorithm, among the four topologies, the longest time it takes to construct effectual route table is no more than 1000s. Consulting the “Completion” column, it can be found that probabilistic flooding algorithm performs better than simple flooding algorithm, but inquiring flooding algorithm does the best.

As usual, transmitting consumes the most energy of a buoy. The broadcast times and inquire times indicate the electric energy consumed by transmitting. For all of the four
topologies and three algorithms, it can be said that inquiring flooding algorithm consumes the least electric energy of all. Take the topology in Fig.4(a) as an example, electric energy consumed by the algorithms is shown in Fig.5. It is assumed that consumed electric power of transmitting is 10W and 0.05W for standing by or receiving.

<table>
<thead>
<tr>
<th>Topology</th>
<th>Algorithm</th>
<th>Time(s)</th>
<th>Broadcast times</th>
<th>Inquire times</th>
<th>Completion</th>
</tr>
</thead>
<tbody>
<tr>
<td>Star</td>
<td>Simple</td>
<td>&gt;1800</td>
<td>47</td>
<td>0</td>
<td>1/5</td>
</tr>
<tr>
<td></td>
<td>Probabilistic</td>
<td>&gt;1800</td>
<td>39</td>
<td>0</td>
<td>2/5</td>
</tr>
<tr>
<td></td>
<td>Inquiring</td>
<td>310</td>
<td>4</td>
<td>5</td>
<td>5/5</td>
</tr>
<tr>
<td>Ring</td>
<td>Simple</td>
<td>&gt;1800</td>
<td>43</td>
<td>0</td>
<td>2/8</td>
</tr>
<tr>
<td></td>
<td>Probabilistic</td>
<td>&gt;1800</td>
<td>39</td>
<td>0</td>
<td>2/8</td>
</tr>
<tr>
<td></td>
<td>Inquiring</td>
<td>980</td>
<td>7</td>
<td>14</td>
<td>8/8</td>
</tr>
<tr>
<td>Cellular</td>
<td>Simple</td>
<td>&gt;1800</td>
<td>48</td>
<td>0</td>
<td>1/10</td>
</tr>
<tr>
<td></td>
<td>Probabilistic</td>
<td>&gt;1800</td>
<td>39</td>
<td>0</td>
<td>7/10</td>
</tr>
<tr>
<td></td>
<td>Inquiring</td>
<td>890</td>
<td>10</td>
<td>10</td>
<td>10/10</td>
</tr>
<tr>
<td>Distributed</td>
<td>Simple</td>
<td>&gt;1800</td>
<td>39</td>
<td>0</td>
<td>3/8</td>
</tr>
<tr>
<td></td>
<td>Probabilistic</td>
<td>&gt;1800</td>
<td>35</td>
<td>0</td>
<td>5/8</td>
</tr>
<tr>
<td></td>
<td>Inquiring</td>
<td>800</td>
<td>5</td>
<td>9</td>
<td>8/8</td>
</tr>
</tbody>
</table>

Table 1: Simulation result comparison

Fig.5 Electric energy consumed by three algorithms

5. CONCLUSION

An inquiring flooding self-organization algorithm for underwater acoustic sensor network is proposed. A process of inquiring neighbours of unknown route is appended to simple flooding once. It is proved by OPNET simulation that compared with simple and probabilistic flooding algorithm, inquiring flooding algorithm constructed effectual route table in a shorter time, and consumes less electric energy.

REFERENCES


DSP IMPLEMENTATION OF TURBO EQUALIZATION BASED UNDERWATER ACOUSTIC MODEM

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Abstract: In this paper, the real time implementation of frequency domain turbo equalizer for single carrier underwater acoustic communication is investigated. The whole receiver system is implemented on a floating point DSP TMS320C6727B board. Through careful adjustment, the real time frequency domain turbo equalizer with burst symbol rate of 2 ks/s is realized with relative large margin. The performance of the system is demonstrated by a tank experiment at NTNU. The system works at relatively low SNR. With respect to the phase rotation due to CFO estimation error, a joint turbo phase tracking and equalizer is proposed. Its performance is also demonstrated through the same experiment data with off-line analysis. It improves the system performance around 2 dB and can be easily implemented in real time with respect to current DSP processing burden.

Keywords: DSP, Turbo Equalizer, Underwater Acoustic Communication
1. INTRODUCTION

Recently, the application of turbo equalization for underwater acoustic communication has been actively pursued due to its significant performance gain over traditional non-turbo receiver. In this paper, we investigate the DSP implementation of frequency domain turbo equalizer for single carrier transmission. The implementation is based on a floating point TMS320C6727B DSP development board. Through careful analysis the working load and code optimization, the real time operation with a burst symbol rate of 2 ks/s has been achieved.

There is not much research on DSP implementation of turbo equalizer available. One DSP implementation of time domain turbo equalizer has been reported in [1]. But in [1], the system was assumed to be perfect synchronized and channel state information was already available for the equalizer. The equalizer was operated in time domain while the equalizer coefficients calculation was based on frequency domain. The modem designed in [2] also included the turbo equalizer mode but the turbo equalizer operates in time domain.

The rest of paper is organized as follows. The transmission scheme and principles of receiver design are described in Section 2 and Section 3. The issue of DSP implementation is presented in Section 4. Section 5 presents the tank experiment setup, results and discussions. Finally, Section 6 concludes the paper.

2. TRANSMISSION SCHEME

The information bits \( \{b_k\} \) are coded by rate 1/2 convolution code[1 21 37]. The coded bits \( \{c_k\} \) are then interleaved by a random interleaver. The output bits \( \{x_k\} \) of interleaver are mapped into QPSK symbols \( \{x_k\} \). The mapped symbols are then divided into blocks. At the tail of each block, a block of unique words (UW) are inserted. The duration of UW should be longer than channel delay spread to avoid inter block interference (IBI).

At the receiver side assuming perfect time and frequency synchronization, the received signal in baseband can be expressed as,

\[
y_n = h_n x_n + w_n
\]

where \( y_n \) is a vector which is composed of the received signal at the \( n \) th block, \( x_n \) is the transmitted symbol vector of the \( n \) th block and \( w_n \) is the additive Gaussian white noise vector with covariance matrix \( \frac{s^2}{W} I \). \( h_n \) represents a \( N \times N \) circular matrix whose first column is composed of zero padded channel impulse response \( \begin{bmatrix} h_0 & L & h_{L-1} & L & 0 & 0^T \end{bmatrix} \). The received signal model in the frequency domain is derived by applying discrete Fourier transform on (1). Then we obtain

\[
Y_n = H_n X_n + W_n
\]
where $Y_n$ and $X_n$ are the Fourier transform of received signal vector and transmitted signal vector respectively, $H_n$ is a diagonal matrix composed of channel transfer function and $W_n$ is the Fourier transform of the noise vector.

3. RECEIVER DESIGN

Our design is focused on the receiver part. The diagram of receiver system is presented in Fig. 1.

![Fig. 1 Scheme of receiving system](image)

The core of the receiver is a frequency domain turbo equalizer. The linear frequency domain turbo equalizer for single carrier transmission scheme which uses the external information provided by the decoder as the prior information for the equalizer can substantially improve the performance of communication system. Its performance has been analysed by [3] for radio communication and by [4] for underwater acoustic communication. With respect to (2), the MMSE criteria based frequency domain equalizer vector is computed as

$$P_n^H = \frac{s_X^2}{s_X^2} H_n^H \left( \frac{s_X^2}{s_X^2} H_n H_n^H + s_n^2 I \right)^{-1}$$  \hspace{1cm} (3)

where $s_X^2$ is the mean of the variance of transmitted symbols. The output of equalizer is then expressed as

$$\bar{X}_n = \bar{X}_n + P_n^H (Y_n - H_n \bar{X}_n)$$  \hspace{1cm} (4)

The symbol estimation from frequency domain equalizer is first converted to time domain through IFFT and then transferred to external log likelihood ratio (LLR) for soft decoder. The soft de-mapper is based on the assumption that the symbol estimation can be expressed as

$$\hat{x}_n = \hat{x}_n + g_0 (x_n - \hat{x}_n) + z_n$$  \hspace{1cm} (5)

where $z_n$ is assumed to be a zero mean Gaussian random variable with variance $s_z^2$. The scale factor $g_0$ and variance $s_z^2$ are computed as

$$g_0 = \frac{1}{N} \text{tr} \left( P_n^H H_n \right)$$  \hspace{1cm} (6)

$$s_z^2 = \frac{s_n^2}{N} \text{tr} \left( P_n^H P_n \right) + \frac{s_n^2}{N} \text{tr} \left( P_n^H H_n H_n^H P_n \right)$$  \hspace{1cm} (7)
where $\delta_H^2 = \frac{1}{N} \sum_{i=1}^{N} s_i^2$. To compute the external LLR, we use the following approximation, with respect to the symbol mapper in Table 1.

\[
LLR_{\text{ext}}(s_{k,1}) = \frac{2 \sqrt{2 \sigma_0} \Re(z_j + g_0 g_x)}{s_z^2}
\]

and

\[
LLR_{\text{ext}}(s_{k,2}) = \frac{2 \sqrt{2 \sigma_0} \Im(z_j + g_0 g_x)}{s_z^2}
\]

The conversion from external LLR to symbol mean and symbol variance is omitted here for brevity. The details can refer to [5].

<table>
<thead>
<tr>
<th>CFO</th>
<th>Training symbol</th>
<th>Data block</th>
</tr>
</thead>
<tbody>
<tr>
<td>00</td>
<td>$-\frac{1+i}{\sqrt{2}}$</td>
<td>10</td>
</tr>
<tr>
<td>01</td>
<td>$-\frac{1-i}{\sqrt{2}}$</td>
<td>11</td>
</tr>
</tbody>
</table>

\[ Table 1. QPSK symbol map \]

The carrier frequency offset (CFO) caused by either relative movement between transmitter and receiver or clock drifting can severely degrade the system performance especially for the block transmission system. To compensate CFO, we use the method in [6] to get an initial estimation of carrier frequency offset. Three identical training data blocks are transmitted at the beginning of each transmission scheme.

The equalizer requires the channel state information (CSI) to compute the coefficients. This is realized by inserting training sequence at the beginning of each data frame and the CSI is estimated by least square (LS) method as

\[
\hat{h} = \left( x_{tr}^H x_{tr} \right)^{-1} x_{tr}^H y_{tr}
\]

where $x_{tr}$ is the matrix of training symbols and $y_{tr}$ is the received signal vector with respect to the training symbols. The CSI is further tracked block by block in the decision directed manner as following.

\[
H_k = \frac{1}{H_{k-1} + (1 - l) \frac{y_k}{x_k}}
\]

The frame structure is shown in Fig. 2.

4. DSP IMPLEMENTATION

The receiver system is implemented on a DSP development board. The development board includes a floating point DSP processor TMS320C6727B which operates at 350
MHz. The system is implemented with C language for the purpose of speeding up development process and code compatibility. The level 0 optimization is used to improve code execution efficiency. The data is processed in a ping pong mode. Two data buffers are set in the external SDRAM. The sampled data from AD converter (ADC) is fetched by direct memory access controller (DMA) through McBSP port and stored in these two buffers alternatively. When one of the buffers is full, the DMA will send an interrupt request to the CPU. The CPU starts to process the data collected in this buffer while the DMA continues fetching data into another buffer.

The system operates in two stages. The first stage is to establish time synchronization between transmitter and receiver. The LFM signal is used for synchronization due to its high correlation peak. The first detected significant peak is assumed to be the starting of data transmission.

The next stage is communication stage. The passband signals are first transferred to baseband by IQ-demodulation. The low pass filter used in the IQ-demodulator is implemented by using over-lap-add method. The fractional equalizer with two samples per symbol time is used in our implementation.

The training matrix for LS channel estimation is pre-computed and stored in on-chip memory. The soft decoder is implemented by BCJR algorithm. The backward recursion is executed first and computed metrics are stored in RAM.

5. TANK TEST

The system is tested in the tank at NTNU. The tank is around 4 m long and 3 m wide with a depth of around 2 m. The distance between the transmitter and receiver is around 1 m and the depths of the transmitter and receiver are around 1 m. The symbol rate for the test is 2 ks/s and the block size is 256 with the UW length of 116 considering the possible severe multi-path propagation. Each frame is formed by 2 blocks of training signal and 5 blocks of data signals. In the test, 6 frames are transmitted to evaluate the performance.

The SOUND DEVICE 722 is used to broadcast the audio file. The received signal is amplified, filtered and finally sampled with a sampling frequency of 48 kHz by the ADC on DSP board. The DSP board processes the received signals and sends the decoded bits to laptop through UART port. During the test, a CFO around 0.5 Hz with the carrier frequency at 12 kHz is observed due to the clock drifting between SOUND DEVICE and DSP board.

Since the block duration is 128 ms in this test case, the maximum available processing time for each block is 128 ms. The CPU cycles required for different functions in the receiver are summarized in Table 2. The consumed DSP cycles for turbo equalizer and SISO decoder are counted per iteration. All the processing parts are with the maximum processing time limitation. The most time consuming process in this case is LS channel estimation which is only required to be operated during the training block and no iterative operation is required. And the real time limitation for this process is longer than 1 block duration. Thus the aim of real time processing is achieved.

<table>
<thead>
<tr>
<th>Function</th>
<th>CPU cycles</th>
</tr>
</thead>
<tbody>
<tr>
<td>Equalizer</td>
<td>232808</td>
</tr>
<tr>
<td>MAP Decoder</td>
<td>1619061</td>
</tr>
<tr>
<td>Soft Mapper</td>
<td>147262</td>
</tr>
<tr>
<td>Demodulation</td>
<td>4216695</td>
</tr>
<tr>
<td>LS channel estimation</td>
<td>33189690</td>
</tr>
</tbody>
</table>

Table 2. Number of DSP cycles per function
The estimated channel impulse response (CIR) is shown in Fig. 3. The CIR spans up to around 200 taps. The bit error rate (BER) performance for the first three iterations is listed in Table 3. The BER at 5.3 dB which is relatively low in practical shows very little improvement during iteration process. This is due to the error of CFO estimation which results in the burst error during transmission. Further improvements can be achieved by joint turbo phase tracking and equalization. The scatterplot of the estimated symbols from the joint turbo phase tracking and equalizer is presented in Fig. 4. The BER for joint turbo phase tracking and equalization is listed in Table 5.

![Fig. 3 Estimated CIR](image1)

![Fig. 4 Scatterplot of Equalizer](image2)

<table>
<thead>
<tr>
<th>SNR</th>
<th>1st Iteration</th>
<th>2nd Iteration</th>
<th>3rd Iteration</th>
</tr>
</thead>
<tbody>
<tr>
<td>No noise</td>
<td>0</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td>6.3 dB</td>
<td>0.09%</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td>5.3 dB</td>
<td>1.48%</td>
<td>1.39%</td>
<td>1.37%</td>
</tr>
</tbody>
</table>

Table 4. BER performance of turbo CFO tracking and equalizer

6. CONCLUSION

In this paper a DSP based real time implementation of single carrier frequency domain turbo equalizer is presented. Its performance is demonstrated through water tank. The real time process could be achieved through careful adjustment during DSP implementation. And the performance is promising at moderate SNR for practical situation. Considering the problem residual error from CFO estimation, a possible solution by using joint equalization and phase tracking is proposed. The implementation of joint turbo phase tracking and equalization will be done in future. Considering the current margin of processing time, real time implementation of joint turbo phase tracking and equalization can be achieved.

<table>
<thead>
<tr>
<th>SNR</th>
<th>1st Iteration</th>
<th>2nd Iteration</th>
<th>3rd Iteration</th>
</tr>
</thead>
<tbody>
<tr>
<td>6.3 dB</td>
<td>0</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td>5.3 dB</td>
<td>0</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td>3.3 dB</td>
<td>0.0913%</td>
<td>0</td>
<td>0</td>
</tr>
</tbody>
</table>

Table 5. BER performance of joint turbo CFO tracking and equalizer

7. ACKNOWLEDGEMENT
The author would like to thank SensIs project from the Research Council of Norway (grand no. 217234) for supporting.

REFERENCES


Orthogonal Multicarrier Underwater Acoustic Communication Experiments in River, Lake and Shallow Sea

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\textsuperscript{b}College of Underwater Acoustic Engineering, Harbin Engineering University. Harbin, China

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\textbf{Abstract:} The purpose of this research is to test the algorithm’s performance of the Orthogonal Frequency Division Multiplexing (OFDM) system in a multi-system prototype for underwater acoustic communication (UAC). The experiments were conducted in three different channels, which were river, lake and shallow sea, in 2011. The prototype used in the experiment included two communication systems: one was spread-spectrum (SS) and the other one was OFDM. The SS was used to transmit control packets and the OFDM was used to transmit data packets. During the experiments, performances of the algorithm used in this system, such as the channel estimation algorithm, channel equalization algorithm, and Doppler estimation and compensation algorithm were investigated, as well as the robustness of this system in different channel conditions. Details of these experiments, including the locations, times, and channel characteristics, will be outlined in this paper. Experiment results show that the OFDM system can work well in different channels with proper parameters. In different channels, the performance of OFDM system is seriously affected by channel variations and Doppler spread. In the complicated underwater acoustic channel, alterable parameters and effective equalization are required.

\textbf{Keywords:} underwater acoustic communication, multi-system, OFDM, outfield experiment
1. INTRODUCTION

OFDM is a new technology in underwater acoustic communication (UAC), and the modem that is special designed for the OFDM system is not very common. Recent years, our group has done some research on the OFDM technology in UAC, and some modem prototypes have been designed for the OFDM communication [1-5]. These modems can run spread-spectrum (SS) system and OFDM system separately or multiple. This paper presents some experiment results of a modem prototype with multi-system: SS and OFDM. SS is use to transmit the control packets which contains the parameters of the OFDM modulation/demodulation, and the OFDM is use to transmit data packets. The experiments were carried out in different regions where the channel condition is different.

The rest of this paper is organized as follows: section 2 detailed the OFDM system and its performance is studied via extensive experiments in section 3, we conclude with useful information of this OFDM system in section 4.

2. SYSTEM OVERVIEW

2.1. OFDM in Underwater Acoustic Communication

In an underwater OFDM system, pre-processing includes channel coding, interleaving, scrambling, sub-carrier modulation, inverse fast Fourier transform (IFFT), guard interval (GI) insertion and PAPR suppression is needed before the data is transmitted [7,8]. And the receive procedure is inverse.

![Fig. 1: Frame structure of the OFDM system](image)

Fig. 1 shows an OFDM frame structure which contains two parts: header and OFDM symbols. In the header, there are two LFM signal and one CW signal. The first LFM signal is used for coarse synchronization while the second one is used for fine synchronization of the frame. The CW pulse in the middle is used to estimate the Doppler factor. The so-called coarse synchronization is to get the timing results by performing correlation between the received LFM and local LFM. The fine synchronization is to get the timing results by performing correlation between the received LFM and a generated LFM which is affected by Doppler Effect. Both of the two kinds of synchronizations are based on the good autocorrelation and anti-Doppler capabilities of LFM.
In each OFDM symbol, cyclic prefix (CP) and cyclic suffix are added, as Fig.2 shows. The transmitted signal of a complete OFDM frame is as Fig.3 shows.

2.2. OFDM modem

OFDM modem is composed of transmitting transducer, hydrophone and sealed cabin, as Fig.4 shows. Power amplifier, battery, receiver, compass and some other components are equipped in the sealed cabin.

3. EXPERIMENT RESULTS AND ANALYSIS

In order to assess the performance of the OFDM system, experiments in different regions are conducted in 2011. System parameters are shown in Tab.1. During the experiments, parameters such as the FFT points, block pilot spacing, comb pilot spacing, and cyclic prefix points were adjusted timely in response to the actual channel situation.
River trail was conducted in Songhua River, Harbin, China, July 2011. The data rate is 1.8 kbps. Ambient environment of the experiment region is as follows: water is 20 meters deep, river is U-shaped and 300-500 meters wide, river bottom is mud, and the flow rate is nearly 2 knots. Fig.5 shows the channel impulse response at different distances and Fig.6 shows the BER curves when use the different FFT length. BER are all calculated before decoding.

### Table 1: System parameters

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Sample rate</td>
<td>48 kHz</td>
</tr>
<tr>
<td>Frequency band</td>
<td>4-8 kHz</td>
</tr>
<tr>
<td>FFT length</td>
<td>9182, 16384 points</td>
</tr>
<tr>
<td>Carrier spacing</td>
<td>2.93 Hz</td>
</tr>
<tr>
<td>Cyclic prefix</td>
<td>62.5ms</td>
</tr>
<tr>
<td>Cyclic suffix</td>
<td>62.5ms</td>
</tr>
<tr>
<td>LFM length</td>
<td>20ms</td>
</tr>
<tr>
<td>CW length</td>
<td>200ms</td>
</tr>
<tr>
<td>Modulation</td>
<td>QPSK</td>
</tr>
<tr>
<td>Channel coding</td>
<td>convolutional code, 1/2</td>
</tr>
<tr>
<td>Pilot mode</td>
<td>block, comb</td>
</tr>
</tbody>
</table>

**Fig.4: OFDM modem Structure**

**3.1. River Trail**

River trail was conducted in Songhua River, Harbin, China, July 2011. The data rate is 1.8 kbps. Ambient environment of the experiment region is as follows: water is 20 meters deep, river is U-shaped and 300-500 meters wide, river bottom is mud, and the flow rate is nearly 2 knots. Fig.5 shows the channel impulse response at different distances and Fig.6 shows the BER curves when use the different FFT length. BER are all calculated before decoding.
At the position of 1km and 2.3km, the Doppler affection was more severe than at 4km, because the relative movement of two ships were different at different distance. So in above mentioned situation, we obtained that the longer of FFT length, the better performance in river trail.

### 3.2. Lake Trail

Lake trail was conducted in Songhua Lake, Jilin, China, September 2011. The lake is almost 50-70m deep with mud bottom. The OFDM modem is located at 6m deep. Experiments were conducted in different distance, 1.5km, 3.4km and 4.3km. Fig.7 shows the channel impulse response on different distances. During this experiment, the wind is strong at 1.5km, however in 3.4km and 4.3km it’s weaker. The received signal to noise ratio was not very low, so the system performance was mainly affected by channel variations. Tab.3 shows the BER at different distance.

<table>
<thead>
<tr>
<th>Distance (km)</th>
<th>Uncoding BER</th>
<th>Coding BER</th>
<th>Data Rate (bps)</th>
</tr>
</thead>
<tbody>
<tr>
<td>1.5</td>
<td>$4.8 \times 10^{-2}$</td>
<td>$1.6 \times 10^{-3}$</td>
<td>2239.5</td>
</tr>
<tr>
<td>3.4</td>
<td>$5.8 \times 10^{-3}$</td>
<td>0</td>
<td>2239.5</td>
</tr>
<tr>
<td>4.3</td>
<td>$5.59 \times 10^{-2}$</td>
<td>$7.4 \times 10^{-4}$</td>
<td>1883.1</td>
</tr>
</tbody>
</table>
3.3. Sea Trail

Sea trail was conducted in the Yellow Sea, October 2011. This experiment was aimed to test the OFDM modem’s long distance communication. Fig. 12 shows the channel impulse response at the distance of 37km. The received time domain signal is shown in Fig. 13. As we can see from the figure, CW signal is obvious while the LFM and OFDM signals are almost submerged in noise.

![Channel impulse response (37km)](image)

![Time-domain waveform of the received signal at 37km](image)

During the experiment, two ships were anchored in 7 degree wind, as a result, the body of ship ups and downs severely. Then the received signal was affected by Doppler shift heavily. Tab.3 gives the statistical of BER at the distance of 37km with different data rates. As we can see, when the BER is lower than $10^{-2}$, the maximum data rate can reach 426.4bps.

<table>
<thead>
<tr>
<th>Data rate (bps)</th>
<th>BER</th>
<th>uncoding</th>
<th>condoning</th>
</tr>
</thead>
<tbody>
<tr>
<td>231.2</td>
<td>5.62×10^{-2}</td>
<td>0</td>
<td></td>
</tr>
<tr>
<td>321.2</td>
<td>6.34×10^{-2}</td>
<td>2.3×10^{-4}</td>
<td></td>
</tr>
<tr>
<td>426.4</td>
<td>5.83×10^{-2}</td>
<td>8.3×10^{-3}</td>
<td></td>
</tr>
</tbody>
</table>

*Table 3: The OFDM demodulation results at the distance of 37km*

4. CONCLUSIONS

Through a series of experiment in different channels, we obtain some conclusions. Firstly, the range of the OFDM modem can reach 30km. Secondly, Doppler will affect the OFDM system severely, effective Doppler compensation algorithm is critical needed. Thirdly, parameters of the OFDM system need to be adjusted timely and automatically according the channel conditions to get a good performance.

5. ACKNOWLEDGEMENTS
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Development/Introduction of the bio-logging system to realize high data recovery rate using acoustic communication

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Abstract: Project to build a new bio-logging system was started in order to elucidate the dynamics of populations and community of the apex predator fish in the open ocean. One of the goals is to enhance the data collection of individual information by utilizing acoustic communication techniques. And ultimately, to build a new bio-logging system using acoustic communication and to complete the evaluation system will be applicable for the open sea. By using this new system, if individuals equipped with the micro data logger are close to each other,
the data recorded in the loggers can be shared by mutual acoustic communication (Inter-individual communication logger). Hence, it is possible to recover data from other data-logger, if it is difficult to recover multiple loggers. In addition, acoustic receiving system designed for various platforms (eg. low-orbit small orbiting satellites (Iridium) and mobile phones), will make this system a multi-platform correspond data receiving and collecting system. By applying different kind of gears (eg. sink-float type that utilizes mooring type, fishing boats, fishing gear, and Argo floats, large biological-mounted) in this multi-platform correspond receiving system, data collection in the open sea will be possible even without recovering the logger itself. For this, a 20-50% decrease in the comparable conventional cost can be expected. Here, we introduce the basic technology and protocols of this newly developed system and report the verification of the method of transmitting the information, and the communication distance.

Keywords: Ultrasonic communication, bio-logging, receiving system, population dynamics

1. INTRODUCTION

To understand the dynamics of the whole ocean ecosystem, it is necessary to quantitatively compare various stages from the lowest primary production producers to the higher-order predators. There are methods existing to monitor the lowest primary production quantitatively in wide area and satellites for remote sensing are widely distributed. However, despite that acoustic remote sensing and measuring fish finders are being widely used, these technologies remain in the observation snapshot manner that depends on research vessel survey. The volume change of top predators, the possibility of active movement in three dimensions and their wide distribution has great influences on the dynamics. In order to understand the quantitative and dynamic aspects of the entire ecosystem, the need of effective methods that enable quantitative observation and monitoring of the behavior of higher-order consumers such as fish is a pressing issue.

In recent years, one of the techniques for direct measurement of behavior of higher consumer using bio-logging is attracting attention (Kitagawa et al. 2000, 2007). This approach is to get environmental and physiological information from the recorders (data logger) attached to the animals. Bio-
logging has evolved to the state where the application and the recovery of loggers on large marine animals are mostly assured. Precision has also progressed and measurements by smaller scale are possible at present.

On the other hand, because many fish move in school, it is necessary to measure the kinetics of the population. Hence, a system that can measure multiple individuals at one time is required. However, it is to do the behavior measured by mass discharge an individual equipped with a logger, you can not only get information of a single individual from the logger of one, due to the limited discharge Quantity Price of logger is expensive, and the number of collected (such as. 2009 Kitagawa et al) such as that little has become a constraint. In order to overcome this bottleneck, an attempt was made to break-through the measurement technology by increasing the efficiency of data recovery (Fig.1).

2. CONCEPT

2.1. Development of the inter-individual communication system

Current data-loggers will only record the measured data of organism behavior, which is attached. Therefore, if an individual each other wearing the logger are close, we will be develop a system that can be shared by mutual acoustic communication data recorded in the logger. This device is a new type of the information and telecommunications between fish, and is epoch-making telemetry device which communicates among two or more fish equipped with a data logger, and records the behavior information on other fish, for example, time, depth of water, water temperature, etc. on the data logger of another fish.

If at least one of two or more fish succeeds in recovery, it will be the device of the dream that did not carry out an idea until now, either that the behavior history of two or more fish becomes solved. In addition, being able to consider various new applications, if correlator ASIC's is used, and carrying out a big contribution to the solution of the ecology of not only the measurement field of underwater acoustic but a sea mammal ecology is that there is no room to suspect (Sasakura et al).

2.2. Development of data receiving system of multi-platform support

Data receiving system compatible with various platforms is constructed. This system is assumed to be a record type and communication type. The communication type, when it is at the receiver around the organisms that attached logger and sent to the platform by the acoustic communication data recorded in the logger and stored. The stored data is delivered to the user of the land by a constant communication cycle. Recording type can obtain data by
recovering the receiver. As the transmission system to the land, will be developed in mind two of satellite communication and mobile phone. Drifting type fixed and is considered as the installation method. In order to enable power saving and maintenance-free operation for a long time, and combined with the like solar power, will be developed, including the size of the system.

3. DATA TRANSMITTING TECHNIQUE

In general, mutual communication protocol begins with sends a transmission request signal and the other party acknowledges it. The communication protocol such, there is no problem in the case of one-to-one, but requires a complex procedure in the case of N-to-N. Therefore, simply was transmitted together with the ID data at regular intervals, the reception, it was just how to just receive it. That is, the method to integrate the receiver with the pinger. However, the transmission method ID of existing, when multiple transmitters are close, it is impossible to identify the ID by interference. So in such a way as follows, we have solved this problem.

The transducer should be high efficiency and smallest size. The layered piezo-electronic element that was recently developed was adopted. But the own resonant frequency is high for small size PZT. Then the method of low frequency operating had been discovered to use a small PZT element. The prototype transducer was using Langevin method. Its resonant frequency was 62.5 kHz and bandwidth was 10 kHz. The resonant frequency related to the transducer length including PZT element (Miyamoto et al 2010). Maximum length sequence signal was generally used to improve S/N ratio and had also wide applications. The system was applied for the 31-bit M-sequence signal using the phase modulation. The four wave numbers are applied to one bit of 31-bit M sequence signal. It was necessary the multi coded M-sequence signal to recognize the individual target. These multi-coded signals should not be interfered. The 32 Gold codes are provided (Sasakura et al).

4. CONCLUSIONS

Bio logging of conventional, has remained to observe the ecology and behavior of the individual wearing the logger. Be measured at the same time the behavior of a plurality of individuals has been difficult (Mitamura et al. 2012). The logger to be developed in this study can be made by a school of fish-population, the individual group and the crowd within the communication one-to-N (s) not only between individuals of one-to-one. We believe understanding of migration realities of highly migratory fish as a group is advanced by leaps and bounds.
In order to obtain the data, thus far, must be collected and logger. Target organisms, limited to such large marine mammals, sea birds, reptiles secured recovery for it. There are restrictions on spatial scales when measurable. The data receiving platform to be developed in this study, in order to recover in acoustic communication only data recorded in the logger, it is not necessary to recover the logger itself. Can be expected that it can complete the recovery for the logger, data recovery rate was about 10% at best so far is increased dramatically.

5. ACKNOWLEDGEMENTS

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REFERENCES


Fig.1 Image of the new concept data logging system
FULL-DUPLEx, RELATIVE CLOCK BASED AND COLLISION FREE PROTOCOL FOR UNDERWATER ACOUSTIC NETWORKS

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Abstract: This paper presents some experimental results of underwater acoustic communication networks both in lake trial and sea trial. Three special designed MAC protocols were introduced and tested: full-duplex based protocol (FD-MAC), relative clock based protocol (RC-MAC) and collision free protocol (CF-MAC). FD-MAC protocol was implemented on a full-duplex modem, vector hydrophone was used to improve the receiving gain, and dual-mode communication and acoustic baffle technology were applied to reduce the local emission interference. Experiment results show that the exposed/hidden terminal problems can be well resolved. RC-MAC protocol was an improved TDMA protocol suited for centralized networks in the underwater environment, neither the global clock synchronization nor the periodically broadcasting of synchronize signal is needed. Net schedule was arrange by the gateway according to the propagation delays and the sub nodes were working on their local clock by mapping the net schedule to their local time line separately. Working slot for each sub nodes can be customized according to the information flow to improve the network throughput and the main node and sub node can be reversal to prolong the network lifetime. Experiment results show that this protocol was easy to be implemented and work well. CF-MAC protocol was designed for the regional underwater observation networks. This protocol avoids channel contentions by using the top-down channel assignment. In this scheme, temporary channel access is assigned by the gateway/AUV using a require-data-send signal (RDS). RDS is used to wake up the specific node and cannot be heard by the other nodes. Two types of acknowledgement signal were designed to shorten the poll duration and save energy. Practicability of this protocol was validated by the experiments. All modems that we used in these trials are designed and implemented by our lab.

Keywords: underwater acoustic communication networks, MAC protocol, full-duplex, relative clock, collision free
1. INTRODUCTION

The past decades have seen a growing interest in underwater acoustic communication because of its applications in marine research, oceanography, marine commercial operations, the offshore oil industry, and the defence sectors [1-3]. With the advances in acoustic modem technology, research in underwater communications today is focusing on the design of the first wireless networks [4]. Challenges of the underwater acoustic environment include a severely limited bandwidth, poor quality of the physical link, and high latency caused by the low speed of propagation [5]. As a result, direct application of the existing medium access control (MAC) protocol designed for terrestrial wireless networks often results in an inefficient use of system resources. To address these problems, some special protocols were designed in this paper and the experiment results of these protocol testing were presented.

The rest of this paper is organized as follows: Section 2 introduces the design and experiment results of our full-duplex based protocol (FD-MAC) and the relative clock based protocol (RC-MAC) and collision free protocol (CF-MAC) are presented in Section 3 and Section 4. We conclude with directions for future work in Section 5.

2. FULL-DUPEX BASED PROTOCOL

The basic idea of the FD-MAC protocol is to separate the communication channel into control channel (ch1) and data channel (ch2) virtually, assign different frequency bands (f1, f2) and different modulate/demodulate approaches (m1, m2) to transmit control packages and data packages. This schematic reduces the interference from bidirectional communication as much as possible. Fig.1 shows the full-duplex communication working schematic. Control packages like RTS, CTS are transmitted in virtual control channel (ch1, f1, m1), while data packages are transmitted in virtual data channel (ch2, f2, m2).

![Fig.1: Full-duplex communication working schematic diagram.](image)

The FD-MAC protocol is very useful in solving the hidden and exposed terminal problems. In this scheme, the nodes can respond to the new visitors while receiving the data packages, details of the hidden/exposed terminal problem solutions can refer to [6].

Sea trial was conducted to validate the engineering feasibility of the full-duplex communication at the Yellow Sea near northeastern China. We designed and developed three full-duplex modems and they were applied in this trial. Direct Sequence Spread Spectrum (DSSS) approach was employed to modulate/demodulate the control packages at 6-8 kHz frequency band, while the data packages were modulated/demodulated using Orthogonal Frequency Division Multiplexing (OFDM) approach at 4-6 kHz frequency band. In trial, these modems were set in a line with a distance of 100 m from each other,
10 m below the surface as Fig.2 (a) shows. Node C was used to simulate the hidden terminal and exposed terminal. We monitored the receiving signal at node B, the received original signal and its spectrum on hidden/exposed terminal conditions are separately shown in Fig.2 (b) and (c). As a statistic result, above 95% of the received packages can be decoded correctly either in hidden terminal conditions or in exposed terminal conditions.

Fig.2: Sea trial (a) setup schematic and monitor records of original signal and its spectrum on (b) hidden terminal condition (c) exposed terminal condition.

3. RELATIVE CLOCK BASED PROTOCOL

RC-MAC protocol is specially designed for the centralized networks. The basic consideration of this protocol is using the local clock to arrange individual work schedule without any complex synchronization, make the nodes transmitting data in parallel according to their propagation delays to improve throughput, and change the gateway dynamically according to its energy level to extend the network lifetime. Work procedure of RC-MAC can be divided into four portions: registration, schedule mapping and data transmitting, clock offset modifying and gateway changing. Details of the RC-MAC protocol implementation can refer to [7].

Lake trial was conducted in April 2013 at Qiandao Lake in Zhejiang province, China. The modem we used in this experiment was developed by our laboratory, as Fig.3 shows. The gateway is deployed near the ship and the 5 sub-nodes were randomly deployed around the gateway in range of 0.5-3 km. Among which, the nearest sub-node was set as the gateway backup. Depth of the experiment region is about 30-50 meters, all these nodes were deployed about 10 m deep. We used FSK to transmit control packets and OFDM to transmit data packets, in frequency band of 6-10 kHz. In schedule registration procedure we use polling registration, it cost about 5 minutes to complete the initialization, then the network start working automatically.

The test was last almost 8 hours and the gateway was changed at about 6 hours after the test started. Statistics of each sub-node sending packets, modifying times and the gateway successfully received packets for each sub-nodes are shown in Tab.1. Where node O is the first gateway and node A is the backup. At first, Node O worked as a gateway and node A worked as a sub-node, when node O’s battery level decrease to 50% of its full level, node O sent out the command to stop the current procedure and make node A work as the new gateway, and itself worked as a sub-node in the rest time. Statistics of node O and node A were when they worked as a sub-node.
Lake trial results are well demonstrated that the protocol is practicable and easy to be implemented, although no clock synchronize was made and no synchronization signal was periodically broadcasted, the network can work well without collision and modify the clock offset automatically during data transmitting. Additionally, the network is energy sensing, which can prolong the network lifetime.

### Table 1: Experiment statistics.

<table>
<thead>
<tr>
<th>Node</th>
<th>Send (packets)</th>
<th>Receive (packets)</th>
<th>Modify (times)</th>
</tr>
</thead>
<tbody>
<tr>
<td>O</td>
<td>631</td>
<td>611</td>
<td>30</td>
</tr>
<tr>
<td>A</td>
<td>725</td>
<td>701</td>
<td>25</td>
</tr>
<tr>
<td>B</td>
<td>1179</td>
<td>1130</td>
<td>113</td>
</tr>
<tr>
<td>C</td>
<td>1355</td>
<td>1341</td>
<td>98</td>
</tr>
<tr>
<td>D</td>
<td>1530</td>
<td>1525</td>
<td>76</td>
</tr>
<tr>
<td>E</td>
<td>1425</td>
<td>1403</td>
<td>89</td>
</tr>
</tbody>
</table>

4. COLLISION FREE PROTOCOL

CF-MAC protocol was designed for the regional observation networks, especially the networks of using an AUV to collecting the obtained data. In CF-MAC protocol, channel contention is not needed, its procedure is specified as follows. The data collector (gateway or AUV) sends require-data-send (RDS) to wake up the requested node (data sender), and assign temporary reserved channel access to it. Then the data sender starts to transmit data to the collector. After data transmission is completed, acknowledgement (ACK) will be sent out by the collector. Here, we design two types of ACK, ACK1 and ACK2. If the data has been received correctly, ACK1 will be sent, the currently data sender goes to sleep immediately upon receiving the ACK1. Simultaneously, this ACK1 is also used to wake up the next node to transmit data to the collector. Otherwise, ACK2 will be sent, this ACK2 can just be recognized by the currently data sender, and the erroneous data packets will be retransmit immediately. The details of CF-MAC protocol can refer to [8].

Lake trial was also conducted in Qiandao Lake in April 2013. This trial was set up of one gateway and 4 sensor nodes with the Conductance, Temperature, and Depth sensor (CTD). The gateway was deployed near the experiment station at a depth about 15 m, and the other nodes were deployed randomly in different regions. The deployment depths and ranges of these sensor nodes to the gateway are respectively: node A (12 m, 2.1 km), node B (14 m, 1.6 km), node C (16 m, 2.6 km) and node D (24 m, 2.3 km). Figure 4 shows the picture of modems applied in this experiment. We invited two modulate/demodulate approaches for communication: Frequency Shift Keying (FSK) for control packets transmission and Orthogonal Frequency Division Multiplexing (OFMD) for data packets transmission, their data rate are 315 bps and 1.05 kbps separately. The control packets length is 64 bits and the data packets length is 512 bits. The carrier frequency band is 6-10 kHz.

The experiment was last 6 days from April 14 to 19. Table 2 shows the statistics of “data request times (\(N_{req}\))” and “correctly received data packets (\(N_{rec}\))” for each node per day from April 15 (00:00) to April 18 (24:00).
The transmitted data from the sensor node to the gateway was the water temperature in the different regions where the 4 sensor nodes were located. Figure 5 shows the observed water temperature variations from April 14 (10:00 am) to April 19 (6:00 am).

5. CONCLUSIONS

This paper is a summary of the recent works for our lab in UACN. We designed some special MAC protocols for the UACN according to the application background and carried out some experiments both in lake and sea. Conclusions we obtained from these works are as follows:

(1) The FD-MAC protocol can efficiently avoid the collision of control packages and data packages by using the full-duplex technology. The experiment results show that the nodes can successfully response to the new visitor while receiving data packets. In this scheme, the hidden terminal and exposed terminal problems can be well solved.

(2) The RC-MAC protocol do not need neither clock synchronization nor synchronize signal broadcasting periodically. This protocol makes the TDMA easy to be implemented in UACN. In this protocol, the end-to-end delay is related to the net cycle, in low net-load chose short cycle can get short end-to-end delay and in heavy net load chose long cycle will be better. The energy sense and work slot irregulate division scheme can improve the network throughput and prolong the network lifetime.

(3) CF-MAC protocol avoids the channel access contentions by using the polling mechanism. As a result, we can obtain a stable throughput in high net-load. In addition, the end-to-end delay and energy consumption is not affected by the net-load changes. The design of the two types of ACK can also provide some other good features in case of the data have been received correctly. This protocol is particularly suitable for the applications of using a mobile node (e.g. AUV) to collect the data from the sensor nodes.
6. ACKNOWLEDGEMENTS

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Session 35

Acoustic Modelling
EFFECTS OF INTERNAL WAVES AND THERMOHALINE INTRUSIONS ON SHALLOW WATER ACOUSTIC PROPAGATION IN THE EAST CHINA SEA

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Abstract: Towed CTD measurements taken during the Transverse Acoustic Variability Experiment (TAVEX) in the East China Sea in 2008 are used to examine the individual and combined effects of density-compensated thermohaline variations (spice) and diffuse and propagating internal waves. A section of the CTD data which includes a large travelling internal wave is examined. Covariance matrices are used to examine the structure of the sound speed variations present in the section on scales of 1000, 100, and 10 meters. Propagation over these distances for both the measured sound speed field and a range-independent case is also simulated using a parabolic equation model, and the difference between the results for the range-dependent and independent cases is examined in order to show correlation between the sound speed variation shown by the covariance and the model results.

Keywords: Internal Waves, Spice, Covariance, Parabolic Equation
1. INTRODUCTION

Acoustic propagation in the ocean is greatly influenced by variations in sound speed due to internal waves (tilt) and density-compensated thermohaline intrusions (spice), but the exact effects of these variations are not yet fully understood. The Transverse Acoustic Variability Experiment (TAVEX) was undertaken with the purpose of advancing knowledge in this area. This paper examines some of the data from that experiment with that goal in mind.

TAVEX was conducted in August of 2008 in the East China Sea, about 100 km southwest of Jeju Island off the southern coast of South Korea. Water column data were collected with a towed Conductivity, Temperature, and Depth (CTD) chain, while acoustic signals were recorded using a bottom-mounted horizontal linear array. The signals were produced by a 300 Hz and a 500 Hz source which were placed 34 km and 20 km, respectively, from the HLA. The location of the experiment, the arrangement of the sources and horizontal linear array and the path that the ship towing the CTD chain followed are shown in Figure 1. Bathymetry information was also collected using a depth sounder during the tow, showing a relatively flat bottom that varies from 70 to 85 meters within the tow area, and from 72 to 80 meters along the acoustic axes [1].

The data collected by the towed CTD chain revealed several large propagating internal waves, as well as smaller-scale variation due to diffuse internal waves and density-compensated thermohaline variations (spice). One of the propagating internal waves is present in the sound speed field in Figure 2, starting around 2 km. In order to study the variability in the sound speed field and its effects on acoustic propagation at different scales, this 6 kilometer sound speed field was decomposed into tilt (internal waves) and spice (thermohaline variations) fields, and segmented into 1000, 100, and 10 meter sections which were each analyzed separately.

Figure 1: Location of TAVEX experiment (imagery from Google Earth) and diagram of the TAVEX sources, receiver, and tow path (courtesy of Chad Smith [1]), with the section used for this analysis highlighted in red.
2. TILT AND SPICE

The 6 kilometer section of sound speed data was decomposed into tilt and spice fields following the procedure described by Dzieciuch [2]. First, isopycnals were traced, recording the temperature, salinity, and depth at each point on each isopycnal. Next, these quantities were averaged to produce \( T(\rho) \), \( S(\rho) \), and \( z(\rho) \). The tilt field eliminates sound speed deviations due to density-compensated temperature and salinity variations by replacing \( T \) and \( S \) with \( T(\rho) \) and \( S(\rho) \). The spice field eliminates internal wave displacements by replacing \( z \) with \( z(\rho) \).

Figure 2: Sound speed field measured by towed CTD chain, with sections used for analysis indicated. The left side shows covariance matrices of each of the 100 and 10 meter sections from the more active portion of the field. All x and y axes on the covariance matrices are depth in meters.
3. SOUND SPEED COVARIANCE

Covariance matrices were calculated for the tilt and spice components and the total field for each of the eight segments shown in Figure 2, as well as a 100 m and a 10 m section taken from section A, which are not shown. These segments were chosen in order to compare the effects of internal waves and spice on propagation over multiple scales and varying levels of activity. Section A is relatively calm, while Section B spans a large internal wave depression.

Covariance is a useful way of visualizing the amount of sound speed variation in different segments [3] (although it only makes sense to compare the magnitude of the covariance among segments of the same length). From the results shown in Figure 2 and Figure 3, it is clear that tilt introduces much more variation than spice, even in sections with no significant internal wave activity. The shorter the section, however, the less difference there is between the tilt and spice covariance magnitudes, indicating that the spicy sound speed variations in the area of the TAVEX experiment are much smaller in scale than the variations due to internal waves.

There is generally more tilt variation within the internal wave (as expected), but the same does not hold for spice. Although spice is often created by the passage of internal waves, it is much longer lasting than tilt, and internal waves are extremely common in the area where TAVEX took place. Therefore, there is likely a considerable amount of spice left behind by waves that passed by previously. This particular wave was travelling nearly perpendicular to the tow path [1], so the spice in section A was probably created by a different wave.

Covariance matrices also provide insight into the depths where the variation is most prolific. Tilt appears to be concentrated around 15 meters (the steepest part of the thermocline), with another concentration in some sections around 30 meters. The spice covariance is less systematic in its distribution. In most cases, there is a concentration around 15 meters, similar to tilt. In some sections, there is very little activity in the rest of the water column, while in others, there are also high points at other depths, which can be either localized or distributed over up to 15 meters.

4. ACOUSTIC MODELING

Propagation at 500 Hz through each of the sections was simulated using the Range-dependent Acoustic Model (RAM) Parabolic Equation model [4]. For all sections, range independent bathymetry was used (depth of about 75 meters, which was the average depth in the section that was used), and an omnidirectional source was placed at 64 meters depth, as in the experiment. Propagation through a range-independent environment was also calculated for comparison. The range-independent sound speed profile was created by averaging over the entire 6 km section.

The results for several of the sections are shown in Figure 3. This figure displays the transmission loss and phase at the end of each field (as would be recorded by a vertical line array at the ranges listed) relative to the same calculation for the range-independent case. For example, if TL-TL\textsubscript{RI} is +10 dB, the environment in question produces a signal which is 10 dB lower than the signal produced by the range-independent environment at the same depth.
Figure 3: Parabolic equation results for 1000 m (top), 100 m (middle) and 10 m (bottom) propagation. Left column: Transmission loss and phase at indicated range for range-independent environment. Center column: Covariance matrices. Right column: transmission loss and phase differences between the field in question and the range-independent field. Total, tilt and spice fields are represented by blue, green and red lines, respectively. For transmission loss, positive values indicate a lower signal level compared to the range-independent case.
For the 1000 m and 100 meter cases, there is very little variation in the propagation due to spice, while in the 10 meter cases, the variation for tilt and spice is more similar in magnitude. This is similar to the results for the covariance. Also similar to the covariance, the level of variation due to spice is much more similar between the calm and active sections, while tilt causes much greater variation in the active sections at all scales.

In many cases, depths where there is significant variation in the signal level correspond to depths where there are peaks in the covariance. Notable examples include the two peaks in the total and tilt response between 30 and 40 meters in section B, and the large amount of variation caused by spice between 10 and 20 meters in section A1a.

Some of the cases show less correlation between the model results and the covariance, however. In particular, the 100 meter range shows almost no variation in the simulation above 20 meters, which is where most of the variation in the sound speed field occurs.

5. CONCLUSION

Covariance matrices of tilt, spice, and total sound speed fields can be used to show in what areas and at what depths the greatest variation in sound speed occurs due to internal waves and thermohaline intrusions. The depths with greater covariance also show correlations to depths where a parabolic equation model shows deviation from the range-independent case, but the correlation is far from perfect.

6. ACKNOWLEDGEMENTS

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Abstract: Peak pressure and sound exposure levels of impulsive signals produced in seismic surveys using airgun arrays must be estimated prior to undertaking the survey. The sound exposure level (proportional to signal energy) can be accurately predicted with existing underwater acoustic models if the environmental parameters are known. However, these models do not correctly predict spatial variations of the peak pressure. Empirical analysis of recordings from three seismic surveys in different marine environments was conducted, including (1) a highly range-dependent environment in deep water over the continental slope off South Western Australia, (2) a nearly range-independent environment of shallow water, over a calcarenite seafloor, and (3) a range dependent environment in medium water depth over a layered elastic seafloor. The analysis showed strong correlation between the peak pressure level and the sound exposure level in all cases, with similar coefficients of linear regression. This paper presents a single linear equation to predict the peak pressure level from the sound exposure level in different marine environments. The difference between the empirical prediction of the peak pressure and the measurements follows a nearly normal distribution of about 0-dB mean and 1.6-dB standard deviation. When measurements are not available, the peak pressure level can be approximately predicted by applying the linear equation to the values of sound exposure level predicted with an acoustic model suitable for each environment. For the deep-water environment, the difference between the predicted and measured peak pressure levels is also nearly normally distributed with the mean and standard deviation of 0.09 dB and 2.79 dB respectively. The peak pressure levels predicted directly by a parabolic equation model exceeded the measured values by more than 10 dB at distances greater than 48 km. Therefore, this empirical equation offers a significant improvement in estimating the peak pressure level of airgun signals.

Keywords: Peak pressure, modelling, airgun, empirical equation, impulsive signal, range-dependent environment.
1. INTRODUCTION

The difficulties related to the prediction of spatial variations in the peak pressure level ($L_{\text{peak}}$) of airgun signals in range dependent environments were discussed in [1]. The expected sound exposure level ($SEL$) and $L_{\text{peak}}$ are commonly estimated before the commencement of a seismic survey, to prevent potential impacts of airgun impulsive noise on marine animals. $SEL$, which is proportional to the signal energy, can be accurately predicted with the existing underwater acoustic models when the environmental parameters are known. However, these models are not capable of predicting $L_{\text{peak}}$ in a sufficiently accurate way.

To address this problem, airgun signals from different surveys were analysed. The results showed that the correlation between $SEL$ and $L_{\text{peak}}$ could be approximated by a linear regression. Furthermore, the coefficients of the linear regression were very similar for surveys conducted in different marine environments. This finding motivated a common analysis of all measurements in order to obtain a global set of linear regression coefficients.

This paper presents a global empirical equation, which can be used to predict $L_{\text{peak}}$ of airgun signals in different marine environments with a better accuracy compared to predictions obtained by exclusive use of underwater acoustic propagation models.

2. SEISMIC SURVEY DATA

Data from three different seismic surveys were used in the analysis. They include measurements from ten different tracks in total. The environments are summarised below, and references are included for more detailed descriptions.

1. A highly range-dependent environment in deep water over the continental slope off Cape Leeuwin, in South Western Australia. Measurements were taken from approximately 17 to 94 km from the source, at depths from approximately 1130 to 1740 m and the average depth was 1400 m. The seafloor was modelled as sand, based on results of geoacoustic inversion given in [2]. Data from one track were analysed.
2. A nearly range-independent environment in shallow water over a calcarenite seafloor on a limestone basement close to Dongara in Western Australia [3]. Recordings were made from 1 to 15 km from the source, and the average depth was close to 40 m. Data from six tracks were studied.
3. A range-dependent environment in medium water depth over a layered elastic (calcarenite) seafloor in Bass Strait, between Tasmania and Victoria [4]. Signals were measured from 2 to 13 km, and the average depth along the tracks was 130 m. Data from three tracks were studied.

3. EMPIRICAL PREDICTION

For each individual track, the correlation of $L_{\text{peak}}$ and $SEL$ was analysed to derive linear regression coefficients in all cases. Thus, ten specific linear equations were obtained, with specific regression coefficients $A_i$ and $B_i$ for the slope and offset respectively.
The similarities between corresponding coefficients from different tracks and surveys prompted a global analysis of all the data. All measurements of $L_{\text{peak}}$ were plotted against the corresponding $SEL$. The global data set also exhibited a linear dependence, allowing us to empirically predict $L_{\text{peak}}$ from $SEL$ with the following linear equation

$$L_{\text{peak}}^{\text{emp}} = A_G \cdot SEL + B_G \text{ dB re 1 µPa}$$

(2)

where $A_G = 1.21 \text{ dB re 1 µPa/ dB re 1 µPa}^2.s$ and $B_G = -20.1 \text{ dB re 1 µPa}$ are the global regression coefficients. The root-mean-square residual of this linear approximation is 1.6 dB re 1 µPa.

This single global equation is applied to empirically predict $L_{\text{peak}}$ from $SEL$ in all environments and tracks studied. The plots are presented in Fig. 1 (for three selected tracks). It can be seen that the prediction of $L_{\text{peak}}$ with the global equation (yellow) is nearly as accurate as that from the linear regression equation estimated separately for each environment (green).

Fig. 1: Measured sound exposure level (red), measured peak pressure level (blue), empirical prediction of peak pressure level with the specific regression coefficients (green), empirical prediction of peak pressure level with the global regression coefficients (yellow) for Cape Leeuwin, Dongara and Bass Strait tracks.
The difference $\Delta L_{\text{peak}} = L_{\text{peak}} - L_{\text{emp-Gpeak}}$, where $L_{\text{peak}}$ is the experimental measurements of the peak pressure level, and $L_{\text{emp-Gpeak}}$ is the empirical prediction, was calculated for the global data set and its distribution was analysed. A histogram and a plot of the Cumulative Density Function (CDF) are presented in Fig. 2 along with the normal distribution fit. It can be observed in the histogram that the distribution is slightly skewed, but looking at the histogram and CDF plot, we observe no significant difference between the measurements and the hypothesized normal CDF. Therefore, we could say that a normal distribution with $\mu = 0.00 \text{ dB}$, $\sigma = 1.64 \text{ dB}$ offers a reasonable approximation for the global data set, and hence, it can be used to estimate the probability of $L_{\text{peak}}$ to be less than a certain value.

![Fig. 2: Distribution of $\Delta L_{\text{peak}}$ with all data and fit with a normal distribution. Histogram (left), Cumulative Density Function CDF (right).]

4. SEMI-EMPIRICAL PREDICTION

The ultimate objective of this study is to develop methods for predicting peak pressure levels prior the commencement of a seismic survey (before measurements are available), in order to assess potential impacts of airgun noise on marine animals. However, the numerical models tested do not accurately predict the $L_{\text{peak}}$ values as compared with measurements. Thus, the solution proposed here is the application of the global empirical equation to the values of the $SEL$ predicted with one of the available numerical models of underwater acoustic propagation. The model to be chosen will depend on the specific problem (environment and signal parameters).

The results of the semi-empirical prediction for the data from Cape Leeuwin are presented in Fig. 3. The underwater acoustic propagation model used was RAMGeo which solves the problem using the parabolic equation approximation. This method was found to be the most accurate in predicting the $SEL$ in that particular scenario. The code is based on the Range-dependent Acoustic Model RAM developed by Collins [5], but it also allows layers of sediments to follow the variable bathymetry. Depicted are the actual measurements of $L_{\text{peak}}$ (blue), the prediction obtained directly from the numerical model (red), and the semi-empirical prediction obtained from the numerically predicted $SEL$ and the global linear equation for $L_{\text{peak}}$ (yellow). It can be seen that although the pattern of the semi-empirical prediction is not exactly like the pattern of the measurements, this method provides a closer decay, which implies better results, especially at longer distances, where the simulation significantly differs from the measurements. The difference between the semi-empirical
prediction and the measurements is most likely a consequence of what follows. First, if the actual geoacoustic parameters slightly differ from the geoacoustic model, the prediction of $SEL$ and $L_{\text{peak}}$ with RAMGeo will be different from the measurements. Second, using those predicted values of $SEL$ to predict $L_{\text{peak}}$ will not give the same amplitude of fluctuations for $L_{\text{peak}}$ because those are smaller in $SEL$. However, the decay with range of the semi-empirical prediction is closer to the measurements than the direct prediction with RAMGeo because this propagation model cannot adequately model the effects of scattering from small-scale variations in the environment along the acoustic path.

![Comparison of Peak pressure levels](image)

Fig. 3: Peak pressure level vs range. Measurements (blue); prediction obtained directly with the model RAMGeo (red); and semi-empirical prediction obtained with the global equation applied to the predicted $SEL$ (yellow).

The difference $\Delta L_{\text{peak}} = L_{\text{peak}} - L_{\text{peak}}^*$ was calculated, where $L_{\text{peak}}$ is the experimental peak pressure level and $L_{\text{peak}}^*$ is its semi-empirical prediction with the global equation and $SEL$ predicted with RAMGeo. $\Delta L_{\text{peak}}$ is nearly normally distributed with the mean and standard deviation of 0.09 dB and 2.79 dB respectively. There is a significant improvement over the direct prediction of peak level made with the numerical model. $L_{\text{peak}}$ predicted directly by the parabolic equation model exceeded the measured values by more than 10 dB along some sections of the acoustic path at distances beyond 48 km and varied from 2.5 and 14.6 dB in the range between 70 and 94 km.

5. CONCLUSION

A single global empirical linear equation was obtained from experimental data to predict the peak pressure level of airgun signals from the sound exposure level measured at various distances in different marine environments. The difference between the empirical predictions and the measurements follows a nearly normal distribution of 0-dB mean and 1.6-dB standard deviation.

When no measurements of the signals are available, a semi-empirical method is proposed, whereby the global empirical equation is applied to the $SEL$ values predicted with a suitable
underwater acoustic model. For the Cape Leeuwin environment, and using the parabolic equation code RAMGeo. This linear equation does not predict small-scale fluctuations around the moving-average level, but the probability of the peak pressure to stay below a certain level at a certain distance can be estimated based on a nearly normal distribution of the difference between the measured and predicted values, which has for this particular data set a 0.09 dB mean and 2.79 dB standard deviation.

$L_{\text{peak}}$ predicted directly by the RAMGeo model offered less accurate values, exceeding the measured values by more than 10 dB at certain distances greater than 48 km.

Therefore, the global empirical equation (applied over measurements of $SEL$ or over the values predicted with an underwater acoustic model) offers a good approximation for the prediction of the peak pressure of airgun signals, independently of the environment, and it improves the prediction obtained by direct and exclusive application of an underwater acoustic model.

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REFERENCES


PECULIARITIES OF THE BROADBAND INTERFERENCE PATTERN IN A SHALLOW SEA WITH A SUBSTANTIALLY VARYING BOTTOM RELIEF

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Abstract: The focus of this paper is on the space-frequency interference patterns created by broadband sources in shallow water environments with substantial variation of the bottom bathymetry. The experimental data, discussed herein, were collected in the region characterized by the sharp coastal slope connecting the shallow and relatively deep water areas with the seamounts located in the latter. It was found out that when the source is located in the uniform shallow area, the space-frequency interference structure typical for the shallow sea is formed in the deeper non-uniform area at the certain distances from the slope and the corresponding frequency interval. This phenomenon can be explained by formation of the fairly narrow directivity diagram of low-order modes in the shallow water area, at the shelf boundary. When the source is towed in the deep part of the waveguide, the interference structure formed in the latter is entirely determined by signals scattered by the coastal slope and, only to a minor extent, by the elevations of bottom relief. At that, in the shallow area, the interference structure is formed by signals scattered by the elevations only.

Keywords: Interference pattern, shelf, normal modes
In studies of sound propagation in the ocean, considerable attention is paid to the formation of acoustic fields in oceanic waveguides with a substantially varying bottom bathymetry characteristic for a transition zone from a shallow to deep region [1, 2]. Usually, the spatial dependence of sound intensity at various frequencies is selected as the informative value, which changes are compared with the corresponding variations of the bottom relief with the use of the adiabatic approximation [2] or the parabolic equation solution [3]. However, the space-frequency distribution of intensity \( J(f, r) \) of broadband sound is more informative, since the striations express the mutual interference between modes and provide useful information inter alia for determining the coherence of waveguide modes.

Forming the interference structure of broadband sound in stratified and uniform by range waveguides was studied in detail in [4].

This paper describes the formation of the interference structure of broadband sound in the oceanic waveguide with an abrupt fall (steep) from the shallow sea segment to the deeper basin (see Fig. 1) [5].

Prediction of interference phenomena in substantially range-varying waveguides is a sophisticated problem. However, it follows from general considerations that if the directivity diagram of a point source located in a uniform region of a non-uniform waveguide turns out fairly narrow in this region, then in a deeper non-uniform region the regular interference pattern also can be expected.

Such a narrow diagram is formed, for example, in an isovelocity waveguide with an absorbing bottom due to a greater attenuation of high-order modes relative to low-order ones [6]. However, in a waveguide with a near-bottom sound channel and an absorbing bottom, the minimum value of the attenuation coefficient corresponds to the mode with the certain number \( l = l_v \) significantly different from unity (see [6]). In the considered case, this number corresponds to the maximum value of the spatial interference period of its adjacent modes \( R_{l_v, l_v} \) (or \( R_{l_v, l_v-1} \)) [6], i.e. the Brillouin ray with maximum cycle length (by the distance \( r \)).

To verify this supposition of forming the interference structure in range-dependent waveguides, experimental studies of the sound intensity distribution \( J(f, r) \) were performed in the shallow sea, where the sound speed profile \( c(z) \) smoothly varied with range (Fig. 1a) and the water depth \( H(r) \) had a sharp change (see the steep in Fig. 1b).

![Fig. 1: The bathymetry and sound speed profiles c(z) for r = 0 (1) and R (2).](image-url)
The 20–150-Hz source was towed at the depth of 20 m from the point \( r = 0 \) of the waveguide I (with the sea depth \( H \) changing from 55 m to 60 m) to the point \( R \) of the deeper waveguide II (with more irregular sea floor) (see Fig. 1). The single receiving hydrophones were disposed near the shelf break at the points A and B (the waveguide I) with the depth of 50 m and the 2.5 km distance from each other and the points A' and B' (the waveguide II) at the depths of 40.5 m and 102 m, respectively, and the distance \( y \) of about 8.5 km from the steep.

As is obvious from the frequency-versus-range distributions \( J(f, r) \) and dependencies

\[
J(\mathbf{r}) = (f_2 - f_1)^{-1} \int_{f_1}^{f_2} J(f, r) df,
\]

\( f_1 = 20 \text{ Hz}, \quad f_2 = 150 \text{ Hz} \) (see Fig. 2), up to an exit of the source to the steep, the interference pattern obtained at the points A and B is represented in the form of striations (lines of extreme values of \( J(f, r) \)) with typical for a shallow sea slopes. Change of the slope sign pronounced at low frequencies \( f < f_g = 30 \text{ Hz} \) (\( f_g \) is the Airy phase frequency for the first mode) is due to the different nature of dispersion in the water wave and the bottom wave. Indeed, in the shallow sea, the mode propagation time decreases with frequency in the former, whereas in the latter the propagation time grows with frequency, similar to the dispersion of the water wave in the deep sound channel. Differences in the dispersion properties of the water and bottom waves are reflected in the spectrogram of explosive signal transmitted at \( r = 0 \), the depth of 30 m and recorded at the point A (see Fig. 3).

While the source moves towards the shelf break, the interference patterns at the points A' and B' are similar and differ (from A and B) in certain details (see Fig. 2 c,d), namely, the interference reveals itself only at frequencies above 70 Hz for the point A' and 50 Hz for the point B' (where the source in the waveguide I can produce the rather narrow directivity diagram). Estimate the distance \( x \) from the source to the shelf break at which the narrow diagram can be formed for the homogeneous water layer with the thickness \( H \) bounded by the absolutely rigid absorbing half-space [6]. Obviously, the diagram of transmission from the shallow to deep region will be narrow, if the grazing angles \( \chi_l \) of modes with numbers \( l = [1, L_1] \) are approximate to zero \( \sin \chi_l = \frac{\lambda}{2H}(l - \frac{1}{2}) \approx 1 \), \( \lambda \) is the wavelength of the transmitted signal, whereas the high-order modes \( l > L_1 \) with steeper angles decay at the corresponding distances. Then, the conditions of: 1) forming the low-order modes at the distance \( x = x_0 = 2H^2/\lambda \) and 2) attenuation of the high-order modes by at least a factor of \( e \) at the distance \( x_L \approx \pi^2x_0^2/(L_1 + 1/2)^2\delta H \) give the desired solution

\[
x = \max\{x_0, x_L\}; \quad \delta \text{ is the dimensionless attenuation factor of modes. The obtained diagram has to be so narrow that the beam formed by modes of the waveguide I does not extend through the entire thickness of the waveguide II at the distance } y \text{ (from the steep to the points A' and B'); i.e. the modes of the waveguide II are not formed. This is possible when the frequency range and the interval of distances satisfy the condition}
\]

\[
y \leq y_L = \frac{(H_{\max} - H_{\min})H_{\min}}{\lambda(L_1 - 1/2)},
\]

where \( H_{\max} = \max\{H(r)\}, \quad H_{\min} = \min\{H(r)\} \). As the frequency decreases, the diagram of the point source located in the waveguide I is formed at smaller distances \( x \) from the steep. However, the beam width grows, and the limiting distance \( y_L \) at which the described interference pattern (resulted from coherent summation of modes from the waveguide I) can be observed – decreases. Hence, the interference patterns at the points A, B and A', B' will be similar only at relatively high frequencies; that actually takes place (see Fig. 2).
Fig. 2: Distributions of sound intensity $J(f,r)$ and its range dependence $J(r)$ over the 20–150-Hz frequency band measured at points A (a), B (b), A' (c), B' (d).
Naturally, in real-life environments, sound speed stratification should have a sufficient impact on a field structure in a high-frequency interval. In the given case (see Fig. 1), this results in broadening of the directivity diagram at output of the waveguide I and greater spreading of the mode beam (formed by the waveguide I) in the waveguide II. The latter circumstance takes place because, in the near-bottom waveguide, the minimum value of attenuation factor corresponds to the mode with the number \( l_i > 1 \) [7]. Furthermore, presence of such channel in a waveguide with increasing water depth conditions forming shadow zones near the free surface [6], which arise starting from a certain distance; at that, the faster the bathymetry varies in range, the shorter this distance is. That is why, at the point A', the interference pattern above the frequency of 100 Hz is substantially weaker in comparison with that at the point B' (see Fig. 2 c,d) and noticeably appears only in the narrow interval from 70 Hz to 100 Hz. The absence of interference effects in a broader range of low frequencies \( f \leq 70 \) Hz at the point A' in comparison with \( f \leq 50 \) Hz at the point B' is explained by an influence of the free surface on the interference suppression of sound (in this frequency band) received at the point A'.

Thus, when a broadband point source is located in a uniform region of a generally non-uniform waveguide, an interference pattern characteristic for this regular region can be observed in deeper and essentially varying in range segments (at certain distances and in corresponding frequency intervals bounded above and below).

When the source exits into the waveguide II, the sound intensity significantly decreases at the points A and B (see Fig. 2 a,b) and increases at the points A' and B' (see Fig. 2 c,d). Such behavior of \( J(r) \) is due to the scattering of acoustic waves by the coastal slope, as it was in [8]; so the interference pattern at the points A and B is virtually imperceptible, and that at A' and B' – significantly changes. The latter is due to the interference of the scattered signal and the direct one in the point A' (or B'), whereby, in the region \( r > 14 \) km, lines of extreme spectral level \( J(f,r) \) have opposite slopes (see Fig. 2 c,d), characteristic, e.g., for the interference of broadband sound in the deep sound channel [4].

Note that when the source is towed over elevations of bottom relief in the waveguides I and II, the interference patterns recorded at the points A, B, A' and B' manifest extreme lines \( J(f,r) \) with opposite slopes (see Fig. 2). Like the case of passing over the steep, this is due to the interference of signals scattered by bottom elevations and signals directly transmitted by the source, for which relative time delays always increase with decreasing distance between the corresponding points. As is known [4], such a line structure is typical for homogeneous range-independent waveguides only when a source passes at minimum distance from a receiver. The very fact of interference between the scattered and direct signals in an oceanic waveguide can be used, e.g., in studies of bottom inhomogeneities.

Fig. 3: Spectrogram of the transmitted signal.
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ACOUSTIC REMOTE SENSING OF INTERNAL KELVIN WAVES IN A STRATIFIED LAKE

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Abstract: Internal Kelvin waves (as any kind of internal seiches) may have great ecological significance in stratified water basins, such as lakes, gulfs, bays, since they induce mixing, resuspension and material transport at the basin periphery, affecting chemical regime and ecosystem productivity. Here we suggest acoustic methodology for parameterization of the basin-scale internal waves. In a 40-m deep stratified lake the effects of internal Kelvin waves (IKWs) on spatiotemporal variability of the mid-frequency (0.5 to 1.5 kHz) sound field were studied in numerical experiments. Specifically, we investigated the horizontal shift of the interference structure and frequency shift of the sound field using a single receiver. It was shown that the IKWs cause significant variations of the sound field that can be easily measured using a linear array of receivers (microphones) or even a single receiver. Basic relations connecting interference pattern shifts and IKW parameters were shown. These relations can be utilized for characterization of IKWs in nature.

Keywords: internal waves, Kelvin waves, shallow water acoustics, frequency shift
1. INTRODUCTION

Internal waves (IWs) and associated water motions cause an increased turbulence and diapycnal mixing at the peripheral lake zones, especially in areas where the metalimnion touches the bottom. These processes enhance vertical transport of solutes from the nutrient-rich hypolimnion to the upper productive stratum and thus enhance primary productivity. Interaction of IWs with the bottom at the periphery of stratified lakes may cause multiple impacts on the vertical and horizontal transport of biologically active solutes. These processes may increase microbial activity in the thermocline, the bottom sediments, and the benthic boundary layer. Therefore, knowledge on IW dynamics and behavior is important for understanding and prediction of ecological processes in stratified aquatic ecosystems. In the paper authors propose underwater acoustic methodology for accurate and fast portraying of such complex water movements.

From the acoustic point of view, a lake constitutes shallow-water waveguide for sound propagation, which is similar to waveguide in the ocean’s shelf zone. In lakes, spring-summer formation of a sharp thermocline zone, where vertical temperature gradient can reach 1–4 °C per meter, and thus, affects the sound speed profile and a distinct near-bottom sound channel (Fig. 1b).

For the purpose of acoustic sensing of lakes, a “mid frequency” (1–10 kHz) range seems to be optimal due to (i) the comparatively small decay of the sound intensity over distances of a few kilometers, (ii) the simplicity of sound signal generation, and (iii) high sensitivity of interference pattern in the waveguide to fluctuations of the thermocline depth that can be determined using acoustical measurements.

Here we consider the effect of the internal Kelvin wave (IKW) on sound field in a stratified lake. The main goals of this paper are (i) to model the response of the sound field to IKWs and (ii) to show how IKW parameters can be reconstructed on the basis of the record of the sound field fluctuations.

2. GEOACOUSTIC MODEL OF LAKE

As an example we consider sound signals propagating in Lake Kinneret (Israel) along an acoustic track AF (Fig. 1) from the source at stn. A (~42 m water depth) to the receiving system at stn. F (~22 m water depth). Stns. A and F coincide with the locations of thermistor chains during the field measurements carried out from June 28 to July 1, 2011. The unperturbed waveguide is shown in Figure 1, and the typical bottom parameters are, as follows: sound speed $c_b = 1600$ m/s, density $\rho_b = 1800$ kg/m$^3$, attenuation coefficient $\alpha = 0.33$ dB/\(\lambda\). The vertical displacement $\zeta$ due to IKW in the cylindrical coordinate system $(r, z, \phi)$ with the center at stn. A, where the z-axis is directed upwards, is:

$$\zeta(r, z, \phi, T) = -(\zeta_0 r / r_b) \cos(\Omega T - \phi) \Psi_1(z)$$

(1)

Here $T$ is time referring to diurnal evolution of the IKW, unlike $t$ is the time referring to variability of the sound field with frequency $\omega$, $\Psi_1$ is the first gravitational mode of IKW, and $\zeta_0$ is the IKW amplitude at the range $r_b$. $\Omega = 2\pi f_{IKW}$ is the Kelvin-wave frequency, in the present case $\zeta_0 = 5$ m, $r_b = 6000$ m, $f_{IKW} = 1/24$ cph.
Variation of the sound field due to IKW dynamics is considered in a frozen time approximation. For the calculation of acoustic amplitude $P(r,z,\phi,T)$ formed by a sound source with frequency $\omega$, placed at point $A(r = 0, z = z_s)$, we use a modal decomposition:

$$P(r,z,\phi,T) = \sum_{m} \psi_m(z,s,T)\psi_m(r,z,\phi,T)\exp\left\{i \int_{0}^{r} \xi_m(r',\phi,T)dr'\right\}, \quad (2)$$

where $\psi_m$ and $\xi_m$ are waveguide modes and propagation constants, depending on time $T$, range $r$, and azimuthal angle $\phi$ as on parameters.

### 3. NUMERICAL MODELING

Sound field amplitude $|P(r,z)|$ in a vertical plane along the transect AF is calculated using eqn. (2) at 1-kHz frequency. $|P(r,z)|$ for different phases of IKWs is shown in Fig. 2 up to a distance of 6 km. Note that at a distance beyond 4 km, only 5 to 6 of the lowest modes contribute to the sound field due to attenuation in the bottom. One can see that the sound field is sensitive to the displacement of the thermocline; an especially strong response can be seen in the peripheral areas, where the thermocline touches the bottom. The movement of interference pattern in a horizontal plane along the acoustic track has a periodical character. The prominent features of acoustic field evolution is that the dynamics of interference pattern below and above the thermocline showing sound field shifts in opposite directions. When the thermocline is moving down, the interference pattern above the thermocline is shifting toward the shore, while the sound field below the thermocline is shifting toward the lake center, and vice versa. Maximal shift of the interference pattern in a horizontal direction is about 600–700 m over 12 hours, such that the average velocity of the motion of the maxima along the HLA is about 0.8–1 m/min, which can be easily registered.

The relationship between displacement of interference fringes (e.g. minima) along the horizontal array (denote as $\delta r$) and the vertical displacement of the thermocline $\delta \xi$ at the distance $r$ from the source A can be derived using the concept of interference invariant:
\[
\frac{\delta r}{r} \approx \frac{\delta \zeta}{H_s},
\]

where \(H_s\) is the waveguide depth at the location of the sound source. Equation (3) is the key relationship that allows estimating the vertical thermocline displacements \(\delta \zeta\) caused by IKWs based on the interference pattern shifts \(\delta r\). In the case of Lake Kinneret, the equation (3) suggests that the horizontal displacement of interference minima (or maxima) is by two orders of magnitude greater than the vertical displacement of the metalimnion. This determines high sensitivity of the acoustical method for parameterization of dynamics of various basin-scale IWs.

Fig. 2. Sound field amplitude along the transect AF at different times. The dark blue area denotes the position of the bottom; the magenta line denotes the thermocline location. The source of 1 kHz sound was positioned near the bottom at a depth of 40 m (stn. A).

Let’s consider a sound interference pattern in a horizontal plane along an acoustic track AF in more detail. Scales of spatial variability of the interference structure in peripheral area \(\Delta_{ml}\) can be provided by interference beating of waveguide modes with numbers \(m\) and \(l\), and determined by a difference of eigenvalues: \(\Delta_{ml} = 2\pi / |\xi_m - \xi_l|\). Quantitatively \(\Delta_{ml}\) is about 50–100 m along the acoustic track. This interference pattern can be registered by a horizontal line array (HLA) with some intervals (e.g. 20–30 m) between hydrophones and total length of at least one order of magnitude larger than \(\Delta_{ml}\) and can be positioned preferably along the acoustic track.

Figure 3 demonstrates the sound interference pattern variability along the range of 400 m, at a 10-m depth above the thermocline (Figs. 3a, b), and on the bottom, below the thermocline (Figs. 3c, d) for two distances from source A: 5600–6000 m and 3600–4000 m, respectively, during one period of Kelvin wave (24 hours). Note that the character of the interference pattern at a 10-m depth (Figs. 3a, c) differs from that at the near-bottom location (Figs. 3b, d). Specifically, for the upper 10-m location the shifts of interference pattern for ranges of 3600–4000 m and 5600–6000 m are rather similar. However, the sound field near the bottom displays mutually opposite shifts of interference fringes (Figs. 3c, d). The last feature occurs due to the exact falling of bottom receivers in the two different water strata: the warm upper stratum (the epilimnion) and the cold lower stratum (the hypolimnion), respectively (Fig. 2c). Therefore, it is possible to detect the point where the thermocline touches the bottom. Beyond this point, all refractive modes become reflected, and the whole interference pattern moves into one direction (Figs. 3a, c). As
mentioned above, the dependence of the thermocline displacement $\delta \zeta$ on variations of $\delta r$ of the sound field interference structure (Eqn. 3) can be in practice used to estimate the dynamics of the thermocline based on acoustic measurements, as it is shown in Figure 4.

Displacements of the thermocline, denoted by stars on Figure 4, were reconstructed using shifts of an interference minimum in the area of 3600–4000 m at different time intervals. As one can see, the estimated thermocline fluctuation is in perfect agreement with the input model of perturbation (Eqn. 1).

Alternative approach to monitoring of the IKW can be based on interference of the pattern variability in the frequency domain. Let’s take position of a single hydrophone at some distance from the source, radiating wide band signal $f(t)$ with a uniform spectrum $S(\omega)$ in the band 800-1000 Hz. The source is placed at the point A (Fig.1). The distances from the source to the receiver (acoustic track AF) are 4 km and 6 km, respectively. Parameters of received signals differ from radiated ones and change during 24 hours (one IKW period) of variation of waveguide properties. One of the most important features of these variations is a frequency shift of interference pattern, or more specifically, a shift of interference maximum. In the Fig.5, temporal variations of amplitude $|P(r,z,\omega)|$ of the sound field are demonstrated. Frequency shifts $\delta f$ can be clear seen on this figure. Using the waveguide invariant theory and (3) we can derive the value of $\delta f$ in relation with thermocline displacement $\delta \zeta$

$$\frac{\delta f}{f} \approx \beta_r \frac{\delta \zeta}{H_s}$$

(4)

Here, $\beta_r$ is the local waveguide invariant at the range $r$. For example, in the considered example it is equal to 6 at the range of 4 km. The values of $\delta f$ estimated using (4) are shown in Fig.5d by blue lines (compare Fig.3d and Fig.5d). Rather good agreement between modelled interference pattern variation and estimated frequency shifts is obtained. Let’s consider the variations of received spectrum in more detail in Fig.5d. Note that temporal variability of received spectrum depends on phase of the IKW $\Phi=\Omega T$. For fixed sound frequency (will receive frequency-filtered signal), for example 800 Hz, over 24-hour IKW period about 11-12 fluctuations with the period of about 2 hours is detected.

Fig.3. Interference pattern variation along the transect AF at a depth of 10 m (a, b) and on the bottom (c, d). Blue lines in (d) denote the interference pattern shifts $\delta r$ calculated using eqn. 3. Vertical axes show distances from the source.
Remark that for the same frequency, the phase of these curves would be different for various depths.

Fig. 4. Model of displacement of metalimnion as a function of time (red line) and displacement obtained using acoustical sampling. Acoustical sampling data (shown as stars) were taken from Fig. 3d at the point of $r=3800$ m.

Fig. 5. Amplitude of the sound field received by single hydrophone in frequency band 800-1000 Hz. Distance from the source is 4 km (b, d) and 6 km (a, c), depth of receiver is 10 m (a, b) or equals to the bottom depth of 40 m (c, d).

General features of interference pattern, which can be used for IW monitoring, are:
- periodical variations of interference pattern with long (about 12h) and short (2-3h) periods;
- period of variations decreases with distance from the source to receiver.

4. CONCLUSION

Usage of the mid frequency range ($<10$ kHz) seems to be optimal for remote acoustic sensing of lakes. Scales of measured parameters of the sound field allow registration and estimation of the mesoscale IW motions in stratified water medium.

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ON THE IMPORTANCE OF UNCERTAIN SEA BOTTOM PARAMETERS FOR THE PREDICTION OF PILE DRIVING NOISE

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Abstract: The topic of hydro sound radiation from industrial offshore pile driving has gained a lot of attention lately. The massive increase in constructed and planned offshore wind farms, especially in the North Sea, leads to a new dimension of hydro sound immission both in quality and quantity. For the impact assessment of future farms and the optimization of insulation systems, reliable numerical predictions of the resulting sound pressure levels (SPLs) are needed. In this contribution, a hybrid approach of a finite element (FE) model coupled to a wavenumber integration (WI) approach is used: The complex phenomena in the vicinity of the pile can be modelled very detailed using the FE model, while the prediction of SPLs for large source receiver separations, can be performed efficiently by the WI method. After a brief introduction to the model, a validation by the comparison to measurements in an offshore wind farm is performed. Here, different model environments are used to represent the sea bottom and their effect on the prediction accuracy is investigated. Subsequently, the problem of parameter uncertainties in the context of underwater noise predictions of offshore pile driving is discussed. Looking at the sea bottom, the degree of uncertainty is relatively high, due to the size of the domain of interest. To address this issue, Monte-Carlo simulations are performed, varying the bottom parameters and the resulting SPL, probability distributions in the water column are evaluated.

Keywords: pile driving, underwater noise finite element method, wavenumber integration, parameter uncertainties, Monte-Carlos method
1. INTRODUCTION

The effect of underwater noise caused by offshore pile driving on the marine environment has recently gained increasing intention, due to the massive construction activities for offshore wind farms. The construction of the turbine foundations, i.e. the driving of large steel piles by hydraulic hammers, leads to high sound pressure levels (SPLs) in the water which are potentially harmful, especially to marine mammals, cf. Kastelein et al. [1].

In this context, a numerical model for the reliable prediction of SPLs is needed for two reasons: On the one hand, the acoustic hydro sound pressures to be expected from planned wind farms need to be estimated a priori. On the other hand, the mitigation measures, which are used to reduce the negative acoustic effects, can hereby be optimized to a certain degree without costly offshore tests.

As offshore wind farms are commonly constructed in shallow water areas, the interaction of the sound waves with the soil has a significant effect on the resulting SPLs in the water column. This is aggravated by the fact that a significant part of the acoustic source, i.e. the pile, is submerged in the bottom. Therefore, two different soil representations are investigated in this contribution by a numerical model and its results are compared to measurements. Closely related to this investigation is the problem of the characterization of the soil by measurements and the degree of uncertainty of the obtained soil parameters. To address this problem, the bottom parameters are varied by means of Monte-Carlo (MC) simulations and the results are evaluated.

After a brief introduction to the applied model in section 2, two different representations of the soil are investigated in section 3. Subsequently, the parameters of an exemplary Pekeris waveguide are varied with the MC method in section 4, before summarizing the results and giving an outlook on future investigations in section 5.

2. NUMERICAL PREDICTION OF OFFSHORE PILE DRIVING NOISE

The prediction of SPLs in this context is complicated because of the size of the domain in relation to the wave lengths of interest. Predictions are needed out to ranges of several kilometres, while the emitted signals exhibit significant energy contents up to the Kilohertz regime. Therefore, a global model, for example by means of finite elements (FE), is impractical due to computational restrictions.

Lately, different models have been developed, using a finite element model for the near field around the pile, coupled to a propagation method for the far field radiation. As an example, Reinhall and Dahl [2] coupled a parabolic equation (PE) model to a transient FE model by means of a point source array along the axis of symmetry of the pile.

For the following investigations, a wavenumber integration (WI) model is used for the far field propagation, which is coupled to a near field FE model in a similar way as the one described above. The spectral components of the single sources are determined from the radiation characteristic of the pile. This is dominated by a quasi-longitudinal impulse which is triggered upon hammer contact and running up and down the pile. The radiated pressures in the FE model are evaluated close to the outer hull of the pile, as the impulse passes by. The moving nature of the source, i.e. the impulse in the pile, is accounted for by
an according triggering of the single sources in the array of the WI model, the moment the pulse would pass their position. For a more detailed description of the model the reader is referred to Lippert and von Estorff [3] or Lippert and Lippert [4].

3. MODELLING OF THE SOIL

The sea bottom transmission path and the interaction of the sound waves at the interfaces have a significant effect on the sound pressure levels in the far field. Therefore, the modelling of the soil is crucial for a precise estimation of SPLs in some distance to the pile. To investigate the needed level of detail of the soil in the model, two different soil representations are modelled in this section, namely a halfspace and a layered soil structure.

As a reference, the simulation results are compared against measurements taken at the offshore wind farm BARD Offshore I in the German North Sea. At this measurement a pile with a length of approximately \(L_{\text{tot}} = 85\) m and a diameter of about \(D = 3.5\) m was driven without sound mitigation. The sound pressure was measured with stand-alone hydrophones at ranges between 30 and 1500 meters in different depths and at different radial positions, cf. itap [5]. For the subsequent investigation the measurements 2 meters above the seafloor are evaluated.

The modelled length of the pile is \(L = 55\) m, of which 40 meters are in the water and 15 meters extrude into the bottom. The material parameters of the soil are chosen in accordance to the geotechnical investigation at the pile driving site, thereby assuming a range-independent environment. This procedure was chosen due to practical considerations, as such survey data is typically available at a very early planning stage of a wind farm. Thereby, it allows for acoustic predictions at the same early stage with no additional cost for the characterization of the bottom.

In the first model, the soil is represented by a homogeneous halfspace. Its material parameters were determined by a layer-thickness weighted average of the first 20 meters of sediment. In the second model, the layered nature of the soil is modelled in accordance with the geotechnical survey. All soil layers were modelled as fluids, with the material parameters taken from Hamilton [6]. The material parameters for the density \(\rho\), the sound speed \(c_p\), and the layer thickness \(\Delta z\) are listed in table 1. Note that the last soil layer is assumed to be a halfspace for both cases.

<table>
<thead>
<tr>
<th>Model I</th>
<th>(\Delta z) [m]</th>
<th>(c_p) [m/s]</th>
<th>(\rho) [kg/m³]</th>
<th>Model II</th>
<th>(\Delta z) [m]</th>
<th>(c_p) [m/s]</th>
<th>(\rho) [kg/m³]</th>
</tr>
</thead>
<tbody>
<tr>
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<td>40</td>
<td>1453</td>
<td>1023</td>
<td>Water</td>
<td>40</td>
<td>1453</td>
<td>1023</td>
</tr>
<tr>
<td>Silty Sand</td>
<td>1.5</td>
<td>1646</td>
<td>1772</td>
<td>Sand</td>
<td>Inf.</td>
<td>1810</td>
<td>2000</td>
</tr>
<tr>
<td>Sand</td>
<td>14</td>
<td>1836</td>
<td>2034</td>
<td>Sand</td>
<td>2</td>
<td>1749</td>
<td>1941</td>
</tr>
<tr>
<td>Sand</td>
<td>Inf.</td>
<td>1836</td>
<td>2034</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Table 1: Material parameters for the two investigated sea bottoms

In figure 1, the SEL and the SPL\(_{\text{peak}}\) are shown versus range for both models, along with the measurement data described above. As can be seen here, both models show a good accordance with the measurement regarding the SEL, especially keeping the
measurement uncertainty of approximately ±2 dB in mind. For the SPL\textsubscript{peak} the degree of 
accordance is lower, as the peak pressure is the more volatile quantity. The oscillation 
around a generally logarithmically decaying trend can be explained by zones of positive 
and negative interference for the broadband source, i.e. the pile. For simple waveguides 
this can be analytically derived using the so-called waveguide invariant $\beta$.

Regarding the comparison of the two models, first of all it has to be noted that the 
results do not strongly diverge. Therefore, the validation with the depicted measurements 
does not yield results strong enough to make a decision about whether the layered 
characteristic of the wave guide needs to be accounted for. In fact, the more simplified 
halfspace model seems to deliver slightly better results for the measurement site which is 
investigated here.

However, for a previous validation with data from a wind farm in the Baltic Sea, the 
layered modelling lead to an improvement of approximately 3 dB, cf. Lippert et al. [8]. 
Therefore, it is concluded that it depends on the peculiarity of the layered structure 
whether it needs to be accounted for or not. To quantify this criterion, further 
investigations at different sites are needed. So far, a layered modelling still seems 
advisable, as for the site at hand the differences where small, compared to the benefit at

4. PARAMETER UNCERTAINTY MODELLING

Even though the models discussed above yielded good predictions for both 
investigated quantities, they are afflicted with a number of uncertainties. For the layered 
model, the material parameters were determined on a relatively coarse basis, whereas the 
simplification was even bigger for the halfspace model. In addition, even if a complete 
survey of the sea bottom was performed, for example by means of geoacoustic inversion, 
the resulting material parameters would still be afflicted with a significant degree of 
parameter uncertainties, see for example Wilken and Rabbel [7]. Finally, as already 
mentioned above, the measurement uncertainty of the hydrosound is relatively large in this 
context, leading to problems in the validation process of the measurement.

To address these sources of uncertainties, the bottom parameters are to be varied in 
this section and the effect on the resulting SPLs is investigated. On the basis of 
geoacoustic measurements at BO I, the degree of uncertainty was assumed to be ±10% of 
the mean value $\mu$, i.e. the standard deviation being $\sigma \approx 0.033 \mu$. 

![Figure 1: Comparison of the simulated SEL and SPL\textsubscript{peak} with measurements for different bottom setups](image)
The parameter distributions are incorporated in the model by means of MC simulations. The basic idea of this approach is to randomly vary and combine the parameters within each of the parameter spaces according to their corresponding probability distribution. For a sufficiently large number of combinations, the results of the quantities of interest converge. The convergence behaviour can be significantly improved by the choice of the sampling method. For the present application Latin Hypercube Sampling (LHS) is used.

Here, as a first step, the soil parameters of an exemplary, shallow-water Pekeris waveguide are varied in the described way and their effect on the SELs in 750 meters distance is evaluated. Each parameter is varied independently, i.e. keeping the other two parameters constant, to determine the sensitivity of the SELs in the water column with respect to each parameter.

The waveguide has a water depth of 18 meters, overlying a sandy halfspace bottom ($c_p = 1740 \text{ m/s}$, $\rho = 1941 \text{ kg/m}^3$, $\alpha = 0.0148 \text{ dB/m}$). The pile has a length of 37 meters. Each of those parameters is varied independently and the SEL is evaluated along the range of the waveguide at a receiver height two meters above the seafloor. Converging results for all varied parameters were obtained using $N = 100$ samples.

In figure 2, the absolute maximum deviation between the minimum and the maximum SEL occurring at each of the receiver positions is depicted. It can be observed that the SEL is most sensitive with regards to the sound speed, followed by the density and the damping. This can be explained by the fact that the sound speed effects both the reflection factor and the critical angle between the water and the soil, whereas the density only changes the reflection factor. The low sensitivity to changes in the damping factors results from the relatively short receiver distance regarded here. As can be seen in figure 2, its influence is linearly increasing with distance and it has been observed to become the dominant sensitivity parameter for simulations out to several kilometres. The oscillating nature of the curves can be explained by a shift of the minima and maxima position, discussed above, for the sound speed and the density.

![Figure 2: Difference between minimum and maximum occurring value \( \Delta \text{SEL} \) within MC simulations for an uncertain sound speed, density and damping](image)

5. CONCLUSIONS AND OUTLOOK
In this contribution, a model for the prediction of offshore pile driving noise was presented, based on a coupled finite element/wavenumber integration approach. Focussing on the long range prediction, different layered soil models were investigated and compared to measurements. Subsequently, the bottom of a Pekeris waveguide was modelled with uncertain parameters using the Monte-Carlo method and the sensitivity of the sound pressure in the water column has been considered.

For the investigated wind farm, the difference between the layered and the halfspace bottom model was negligible. However, previous results suggest that a layered modelling is very important. Therefore, a layered modelling is generally recommended. To get a deeper insight into situations in which this might be neglected, further investigations at different sites are planned.

The sensitivity analysis showed that the sound speed is the most influential factor in the model, followed by the density and damping. Therefore, this parameter should be described with particular care. As a next step, the sensitivity analysis is to be expanded to more complicated, layered waveguide structures. Also, the influence of additional factors, as for example sound speed profiles in the water, needs to be investigated.

ACKNOWLEDGEMENTS

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MODELING PULSE PROPAGATION IN A WEDGE ENVIRONMENT WITH RANGE-DEPENDENT GEOACOUSTIC PARAMETERS

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Abstract: M-sequence coded pulses were transmitted across a slope to a range of 10 km during an experiment in the Florida Straits in 1999. The pulse structures of the long-range received data consisted of secondary arrivals with arrival times consistent with an effect of horizontal refraction from the sloping seafloor. Three dimensional (3D) sound propagation modelling was conducted to characterize the out-of-plane propagation paths for the experimental geometry. A full-waveform pulse simulation confirmed that the secondary arrivals could be attributed to horizontally refracted sound paths. The influence that the seafloor properties had on the characteristics of the out-of-plane arrivals was investigated using a range-dependent geo-acoustic model of the three-dimensional coastal environment. The model was applied to a 3D propagation code and Fourier synthesis was used to simulate the broadband signal structure at 10 km. A geoacoustic model with parameters changing in the upslope direction improved the match of the modelled arrival structure with the measured data.

Keywords: three-dimensional sound propagation, wedge acoustics, pulse propagation, horizontal refraction, range-dependence
1. INTRODUCTION

Underwater sound ray paths can exhibit horizontal curvature when sound propagates over sloping seafloors. The shoreward slope of the seafloor in the Florida Straits causes such horizontal refraction, leading to three-dimensional (3D) out-of-plane propagation effects. This was illustrated through examination of measured and modelled waveforms from pulses propagated along the slope at a location near the South Florida Ocean Measurement Centre [1, 2]. In this paper we examine how the geoacoustic properties of the seafloor also influence this 3D sound propagation using output from a 3D parabolic equation computer model, MONM3D [3].

In 1999 deFerrari et al. [4] collected transmission loss data using a vertical receiving array deployed at 10 km range from a moored deep-water acoustic source. The source and the receivers lay in a plane oriented parallel to the shore with nearly constant bathymetry between 150-160 m depth (Fig. 1). The source transmitted M-sequence coded pulses at centre frequencies between 100 and 3200 Hz. High sensitivity data were received at 13 hydrophones on a vertical receiving array. At frequencies below 800 Hz the received signals consisted of a set of primary arrivals that were followed by a strong set of secondary arrivals with a time separation of approximately 0.4 seconds [4].

In a subsequent geo-acoustic inversion study, using a two-dimensional (2D) forward model, Jiang et al. [1] were unable to determine a geoacoustic model that matched the late arrivals in the received pulse response traces. They speculated that the strong secondary arrivals originated from horizontally refracted sound paths which initially travelled along a shoreward heading. Such arrivals would lag the direct (source to receiver) primary arrivals due to their longer propagation paths. Through vertical beam-forming Jiang et al. showed that the propagation angles contributing to the secondary arrivals were steeper than those for the predominantly low angle primary arrivals. This allowed them to spatially filter the data to shallow arrival angles and to conduct the geoacoustic inversion using only the primary arrival data, assumed to travel in the source-receiver plane.

A 3D acoustic modelling study was later conducted by Sturm et al. [2] to validate the assumption that the propagation paths for the secondary arrivals travelled out of the source-receiver plane. Sturm et al. used Fourier synthesis to reconstruct the waveforms and simulate the transmitted pulses using 2D and 3D solutions to the parabolic wave equation. The 3D calculations did predict secondary arrivals that were absent from the 2D simulations. However the characteristics of the 3D-simulated secondary arrivals did not exactly match those observed in the measured data.

Building upon the work of Sturm et al., the MONM3D model was applied to further investigate the characteristics of the 3D acoustic field at this site [5]. Range-variable geoacoustics were introduced, allowing the properties of the environment to be defined as a function of off-shore range. The computed PE fields were decomposed into their modal components to consider the simulated waveforms in terms of the mode arrival structure to better characterize the components of the late-arriving sound energy.
2. MODEL CONFIGURATION

For this work MONM3D was configured to simulate the Florida Strait environment for an omni-directional point source at a depth of 107 m. Transmission loss was computed for a band of frequencies between 75 and 125 Hz (at 1 Hz resolution) and waveform traces were simulated through Fourier synthesis. The source function used to generate the waveform traces was a Gaussian cosine pulse centred at 100 Hz with a 50 Hz bandwidth.

The sound speed profile input to the model was derived from measurements collected in the source-receiver plane during the 1999 propagation loss experiment [4]. The top of the profile (to a depth of 80 m) was isovelocity, below which there was a strongly downward refracting structure that created a duct at the bottom of the water column for low order modes. The structure of the sound speed profile was assumed to be constant over the modelled region but the profile was truncated from the bottom with decreasing water depth.

Bathymetry data for the model were obtained from the NOAA NGDC US Coastal Relief Model online database. The right panel of Fig. 1 shows the bathymetry profile between the source and the shore. Within 750 m of the source the seafloor sloped toward shore at an angle of 2.5° then the slope increased to an angle of 4° for approximately 1 km before reaching a much shallower slope less than 1° at the near-shore ranges.

A geoacoustic model was configured in which the seafloor parameters were allowed to vary with range in the up-slope direction. It was assumed that coarser sediment could be expected closer to shore at the top of the slope due to natural sediment sorting. This hypothesis was supported by piston core samples collected by researchers at the University of Miami in April 2000 [6]. Two regimes were evident in the cores—near-shore and in-plane—distinguishable by grain size and by compressional sound speed value, with the near-shore cores consisting of coarser, faster sediment.

The environment was thus divided into three geoacoustic provinces. Each consisted of a single homogeneous layer with parameter values listed in Table 1. The province boundaries were selected to demarcate the stepped slope of the seafloor. Province 1 corresponded with the shallow-slope segment; it contained the source-receiver plane and
was assigned geoacoustic properties as defined by Jiang et al. [1]. Province 2 contained the steep segment of the slope, for which coarser sediments were assumed. Province 2 parameter values were consistent with a sediment description in between the types found in the near-shore and in-plane piston cores with decreased porosity and increased grain size compared to Province 1. Finally Province 3 at the top of the steep slope was representative of sediment of very low porosity, with compressional sound speed and density values consistent with the values from the near-shore piston core samples.

<table>
<thead>
<tr>
<th>Geoacoustic Province</th>
<th>$\rho$ [g/cm$^3$]</th>
<th>$c_p$ [m/s]</th>
<th>$\alpha_p$ [dB/λ]</th>
<th>$c_s$ [m/s]</th>
<th>$\alpha_s$ [dB/λ]</th>
</tr>
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<td>1</td>
<td>1.779</td>
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<tr>
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</tr>
<tr>
<td>3</td>
<td>2.1</td>
<td>1950</td>
<td>0.8</td>
<td>490</td>
<td>6.5</td>
</tr>
</tbody>
</table>

*Table 1: Geo-acoustic parameters used in the three provinces of the range-dependent geoacoustic model.*

### 3. MODEL SIMULATION

The resulting waveforms for signals at 10 km range are shown in Fig. 2 for both 2D and 3D model calculations at 13 receiver depths. A red curve marks the envelope of the measured data from deFerrari et al. [4] generated by digitizing the waveforms presented in Sturm et al. [2]. Each trace was normalized to the maximum level for that receiver. The waveforms from the full-3D results contained arrivals at approximately 0.6 seconds that followed the expected direct-path arrivals and were absent from the 2D calculation results. This range-dependent 3D result is in better agreement with the measured data compared to the 2D result.

In-plane propagation paths between the source and receiver travelled wholly within Province 1. Paths travelling slightly out of the source-receiver plane also remained within Province 1, but paths that travelled further up-slope passed through the lossy environment of Province 2. Refracted paths that travelled further up-slope coupled into the source-receiver plane at longer ranges; therefore, the field in the source-receiver plane was influenced at close ranges by paths that travelled only within Province 1 and at further ranges by paths that also passed through Province 2. The sediments in Province 2 resulted in steeper propagating sound paths that refracted more strongly than paths travelling through Province 1. The receivers at 10 km range were in a shadow zone for these strongly refracted paths that would have arrived at times greater than 0.6s.

The PE field was decomposed into its modal components at every kilometer along a track in the source-receiver plane. The shade plots in Fig. 3 present the magnitude of the modes as a function of wavenumber and range in the source-receiver plane for 2D and for 3D model results. The 2D and 3D mode spectra look very similar at close ranges, but begin to illustrate noticeable differences at approximately 6 km range. Whereas the magnitude of individual modes in the spectra from the 2D calculations gradually decreased with range, the spectra from the 3D calculations show modes with increased amplitude at range due to the interference of the different horizontally refracted arrival paths. Also, since horizontal refraction toward the down-slope direction caused the modes to turn out of the source-receiver plane there was a spatial envelope for each mode in the 3D results.
Fig. 2: Simulated waveforms at 13 depths, obtained through Fourier synthesis of PE fields from MONM3D for (left) 2D calculations and (right) 3D calculations with range-dependent geoacoustics.

Fig. 3: Wavenumber spectra as a function of range in the source-receiver plane from PE fields computed using (left) 2D and (right) 3D calculations with range-dependent geoacoustics.

Vertical beamforming was applied to investigate the vertical arrival structure (correspondingly, the mode arrival times) of the modelled waveforms (Fig. 5). The 2D and 3D results were very similar at times <0.4s and both indicated that the first arrivals were composed of low angle vertical beams, that is, low order modes. For times >0.4s the 3D signals contained contributions from out-of-plane modes. The secondary arrivals, denoted by the bright spots in the 3D beam patterns at around 0.6s, corresponded to the propagation angles for modes 6 and 7. The arrival time and duration of this late-arrival signal (Fig. 5, right panel) is in good alignment with the measured data.
SUMMARY

Strong secondary arrivals were observed in simulated waveforms from modelled 3D PE fields. These arrivals were attributed mainly to refracted arrivals of modes 6 and 7. The timing of the simulated second arrivals most closely matched the measured data when the geoacoustic model included range-dependence of the seafloor sediments such that the sediment became more coarse with up-slope range toward shore. Such a configuration removed the propagation paths of modes 6 and 7 that travel up the steep part of the slope but maintained the contribution of paths that did not travel as far up-slope before coupling into the source-receiver plane.

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FINITE DIFFERENCE TIME DOMAIN METHOD FOR ACOUSTIC WAVES IN ATTENUATE AND ABSORPTIVE MEDIUM FOR LAYERED UNDERWATER ACOUSTIC ENVIRONMENTS

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Abstract: In this work, we use Finite Difference Time Domain (FDTD) method to investigate the underwater acoustic wave propagation problem; in order to understand the acoustic wave propagation properties and determining the effects of the attenuation and absorption for the sound intensity in the sea.

The results are presented for calculated Transmission Loss (TL) of propagating acoustic waves in layered underwater acoustic environments by using Finite (FDTD) method. In a first step; the sea environment are modeled as one layered medium before generalizing the resolution to two layered (fluid-fluid and fluid-solid) medium. The sea surface and the sea bottom are considered to be planar surfaces. The discretized acoustic wave equations are used in the algorithm of the FDTD method and Perfectly Matched Layer absorbing boundary condition is applied to eliminate numerical reflections from the ends of the grid .Two cases are discussed (Taking account of absorption and attenuation).The good performance of this method is validated with the results of BELHOP program and by comparison with the analytical method.

Keywords: FDTD, Transmission Loss (TL), Underwater layered, acoustic propagation attenuation, Absorption, PML.
INTRODUCTION:

The main effect of propagation of acoustic waves in water is the decrease of the signal amplitude by absorption. This is due to the chemical properties of sea water, and is a definite factor of the propagating waves. In particular this is a limiting factor to reach high frequencies domain.

The estimation of propagation losses is a very important factor in the evaluation of sonar system performance. The underwater propagation medium is usually limited by two well-marked interfaces: the sea surface (water /air) and the bottom (water/solid).

Consequently, the propagation of an acoustic signal is accompanied by interferences and reflections; which give rise to bursts and replicas of the primary signal at high frequencies and special field of stable interference at low frequencies. These unwanted interferences are common sources of trouble for reception and exploitation of useful signals.

The finite difference time domain (FDTD) approach is widely used as computational method in electromagnetic, capable of computing electromagnetic interactions for problem geometries that are extremely difficult to analyze by other methods[1].

In this paper, the Finite Difference Time Domain (FDTD) Method is proposed to calculate the propagation of sound in shallow water. The recent development of computer system enables the method to be applied in acoustics takes account of attenuation and absorption [2].

For underwater acoustics problems the starting point is the Helmholtz equation, which have many approximations introduced such as: the ray trace [3], FFP (Fast field program) [4], PE (Parabolic equation) [5], or once again the direct numerical solution of Navier stockes (DNS) [6].

For the research environment the parabolic equation PE provides the best compromise between accuracy and efficiency for such problem [7]. The analytical validation is taken from knightly and from Vefring MjØlsne [8]. It involves a homogeneous waveguide of constant depth with a pressure-release surface and a rigid bottom. These parameters yield a maximum of 27 modes which propagate up to about 84° from the horizontal.

In the other hand in order to calculate the amplitude of pressure (P), with respect to depth and range, we use Bellhop program which is a Gaussian beam tracing to find the transmission loss (TL) using an isospeed sound velocity profile [9].We note that the bellhop popular code is especially used in the cases of two layered under water environment (fluid-solid).

In this work The FDTD algorithm is validated both by analytical solutions based on the parabolic equation PE and numerical results provided by the Bellhop program at frequencies as high as 1 kHz for two dimensional problems.

1-FDTD FORMULATION:

A-basic equations:
**Taking account of absorption:**

Two fundamental acoustic equations are numerically solved by using the FDTD method [1] satisfy the basic Newton’s law of motion and equation of continuity:

\[
\rho(r) \frac{\partial v(r,t)}{\partial t} = -\nabla p(r,t)
\]  

(1)

\[
\frac{\partial p(r,t)}{\partial t} + \gamma(r) c^2(r)p(r,t) = - \rho(r) c^2(r) \nabla v(r,t)
\]

(2)

Where \( p(r,t) \) is the spatial and the time dependent acoustic pressure field (N/m²), \( v(r,t) \) is the particle velocity (m/s), \( \rho(r) \) is the density (kg/m³), \( \gamma(r) \) is the absorption coefficient, \( c(r) \) is the sound speed (m/s).

This absorption coefficient is related to the attenuation coefficient \( \alpha(r,\omega) \) by using the complex wave number:

\[
k(r,\omega) = (\omega^2 / c^2 + i\omega \gamma)^{-1/2} = \omega / c'(r,\omega) + i\alpha(r,\omega)
\]

\( c'(r,\omega) \) is the dispersive wave velocity.

**Taking account of attenuation:**

The basic equations of the FDTD method, which is taking account of attenuation, are given as follows:

\[-\rho \frac{\partial v}{\partial t}(r,t) = \nabla p(r,t) + \eta(r)v(r,t)\]

(3)

\[-\frac{1}{K} \frac{\partial p}{\partial t}(r,t) = \nabla v(r,t)\]

(4)

With \( \rho(r) \), \( p(r,t) \) and \( v(r,t) \) are defined as in the first case (taking account of absorption).

\( K \) is the bulk modulus, \( \eta(r) \) is the resistance coefficient that is proportional to the particle velocity and must be decided to take account of the influence of absorption by the media.

\[
K = \rho(r) c^2(r)
\]

\[
\eta(r) = \frac{2\gamma_1\gamma_2}{\sqrt{\gamma_1^2 - \gamma_2^2}} \rho(r) C(r)
\]

\( \gamma \) is the wave number.
Is the attenuation constant in dB/km \([10]\).
\[
\gamma_2 \approx 3.3 \times 10^{-1} + \frac{0.11 f_{\text{kHz}}^2}{1 + f_{\text{kHz}}^2} + \frac{44 f_{\text{kHz}}^2}{4100 + f_{\text{kHz}}^2} + 3.0 \times 10^{-4} f_{\text{kHz}}^2
\]

B- Discritization principal:

The FDTD equations derived from (1) to (4) are discritized in time and space to obtain the update equations given as follow:

**Absorption case:**
\[
\frac{\partial v_x}{\partial t} = -\frac{1}{\rho} \frac{\partial p}{\partial x}
\]
\[
\frac{\partial v_y}{\partial t} = -\frac{1}{\rho} \frac{\partial p}{\partial y}
\]
\[
\frac{\partial p}{\partial t}(x, y, t) + \gamma(r) c^2(r) \rho(x, y, t) = -\rho(r)c^2(r)(\frac{\partial v_x}{\partial x} + \frac{\partial v_y}{\partial y})
\]

**Attenuation case:**
\[
\frac{\partial p}{\partial t}(x, y, t) = -K(\frac{\partial v_x}{\partial x} + \frac{\partial v_y}{\partial y})
\]
\[
\frac{\partial v_x}{\partial t} = -\frac{1}{\rho} (\frac{\partial p}{\partial x} + \eta(r)\nu(x, t))
\]
\[
\frac{\partial v_y}{\partial t} = -\frac{1}{\rho} (\frac{\partial p}{\partial y} + \eta(r)\nu(y, t))
\]

2-NUMERICAL CONSIDERATIONS:

The FDTD cell size is \(\Delta x = \Delta y = \lambda / 20 = 0.9375m\) are chosen for 80 Hz frequency (\(\lambda\) is the wavelength) in the construction of the spatial FDTD grids for evaluation of the geometrical problem details, properly with the time step \(\Delta t = \Delta x / (2c)\) (where c is the speed of sound in the water).

The real physical dimensions of the problem space are 60m x 1000m. And the total number steps are 4000.

The bottom layer is in 20 m of depth for fluid-fluid and 40m depth for the fluid-solid media and all of the boundaries except pressure-release sea surface are modeled by PML absorbing boundary condition. Where the source is placed at the 2m depths as shown in the figure1:

Due to temperature and pressure variations, the velocity of acoustic waves varies specially in the sea, mostly with depth and the paths of acoustic waves are thus refracted depending on velocity variations. In this FDTD simulation, for simplicity reasons, we use isospeed sound velocity profile
Fig.1: Geometrical of the underwater propagating medium

3-RESULTS AND DISCUSSION:

- Taking account with attenuation:

  a) one layered medium
  b) two layered (fluid-fluid),
  c) two layered (fluid-solid).

Fig.2: Acoustic FDTD taking account with attenuation, a) one layered medium

b) two layered (fluid-fluid), c) two layered (fluid-solid).
*acoustic field distribution for the fluid solid two layered environment taking account with attenuation:

Fig.3: acoustic field distribution for the fluid solid two layered environment taking account with attenuation

- Taking account with absorption:

Fig.4: Acoustic FDTD taking account with absorption, a) one layered medium. b) two layered (fluid-fluid), c) two layered (fluid-solid).
CONCLUSION:

The FDTD method is used to calculate the propagation of acoustic plane wave in shallow water with different environment. The simulation results of the transmission loss pattern are compared with analytical results of parabolic equation (PE) [8] and numerical results of Belhop code.

The FDTD is also able to visualize the propagation of the sound field in the same simulation model. We can clearly see the interference between direct wave and reflected waves. It is useful for us to recognize the sound propagation not only in water but also in sediment. These results show the validity of the FDTD method.

Three cases of one layered, fluid-fluid and fluid-solid two layered medium was treated. We demonstrate that our results agree well with Belhop method for all cases at high frequencies. Bellhop was chosen for this analysis as it has proven to be an accurate modeling tool for high-frequency (>1 kHz) transmissions. At low frequencies our Simulation Show also a good agreement with analytical results. We have separately investigate the effect of attenuation related to water viscosity and the dissipative effect related to the absorption coefficient depending on the chemical composition of sea water.

As a future work, it is aimed to develop FDTD model with more realistic properties of the medium like roughness, temperature and pressure for calculating the TL in more realistic environments using spherical acoustic waves in three dimensions space and extend our study to more realistic sound source.
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SOUND FLUCTUATIONS IN THE PRESENCE OF NONLINEAR INTERNAL WAVES MOVING ALONG ACOUSTIC TRACK IN SHALLOW WATER

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Abstract: Theoretical analysis and numerical modelling is carried out for low-frequency (<1kHz) acoustic signals propagating in shallow water waveguide in the presence of nonlinear internal wave traveling approximately along acoustic track. Fluctuations of the sound amplitude at the receiver in this case are provided by modes coupling (in modal theory) or ray scattering (in ray approximation). Modelling is carried out within the framework of modal theory and PE approximation. It is shown that in spectra of amplitude fluctuations there are characteristic frequencies, proportional to speed of internal waves. Maximal amplitude of fluctuations in spectrum takes place at the so called predominating frequency, determined by the scale of interference beating of modes, having the turning point in area of thermocline (or by the cycle of rays, touching thermocline). Depth dependence and frequency dependence of fluctuations is studied.

Keywords: internal waves, modes coupling, shallow water acoustics
1. INTRODUCTION

The influence of nonlinear internal waves (NIWs) on the sound field in the ocean has long been a subject of research. This paper considers sound field fluctuations resulting from mode interaction for the propagation of low-frequency sound that intersects the fronts of NIWs of significant amplitude (~10 m) and moving at a typical speed (~1 m/s). As we will show, the sound field fluctuation frequency caused by NIW motion have values of \( F < 10 \text{ c/h} \), which is close to oceanographic values, e.g., the Vaisala frequency.

In the authors' previous works the analogous problem of sound and NIW interaction was considered experimentally (based on the SW06 experiment) using the ray approximation. It was shown that fluctuations have a quasi-periodic character with a dominant fluctuation frequency of \( \nu^* = \nu/D \), where \( \nu \) is the velocity of NIWs along the track, and \( D \) is the ray cycle (called critical), which has the maximal interaction with NIWs and makes the largest contribution to intensity fluctuations.

2. MODE THEORY OF FLUCTUATIONS

Let us consider sound propagation in the vertical plane passing through the source and receiver (Fig. 1). We select the origin on the upper surface of a shallow-water waveguide above the source, we direct axis \( z \) vertically downward, and axis \( r \), horizontally toward the receiver.

In the absence of NIWs, we assume the waveguide to be plain-stratified, with depth \( H \) and a sound velocity profile of \( c_0(z) \). In the presence of NIWs, the sound velocity profile has the form \( \tilde{c}(r, z, T) \), where dependence on time \( T \) appears due to NIW motion.

\[
\tilde{c}(r, z, T) = c_0(z) + \delta c(r - \nu T, z),
\]

where \( \delta c \) is perturbation of the sound speed profile and \( \nu \) is the velocity of NIWs along the acoustic track. Quantity \( R = \nu T \) characterizes the position of the back front of NIW from the horizontal axis (Fig. 1) if for \( T = 0 \) it was located over the source. For convenience of calculations, we assume NIWs to be localized in a certain region \( R < r < R + L \), where \( L \) is the size of NIWs along the track.

Quantities \( \nu \) and \( L \) depend on angle of inclination \( \alpha \) of the NIW front to the track in the horizontal plane: \( \nu = \frac{\nu_\perp}{\sin \alpha}, \quad L = \frac{L_\perp}{\sin \alpha} \), where \( \nu_\perp \) and \( L_\perp \) are the velocity and horizontal dimension of NIWs perpendicular to the front. Due to the slowness of NIW motion, the sound field is calculated for each fixed value \( T \) (approximation of a frozen
medium or "slow" time). Suppose that \( k = \omega / c \) is the wavenumber for some mean sound velocity \( c \), \( n = c / c_0 \) is the refractive index of the medium in the absence of NIWs, \( \mu = c^2 \left( \frac{c^2}{c_0} - 1 \right) \approx -2 \delta c / c \) is an addition to \( n^2 \) caused by the presence of NIWs. Owing to this, \( \mu(r, z, T) = \mu(r - \nu T, z) \). In the absence of NIWs, \( (r < R) \) we have \( \mu = 0 \) and solution of the Helmholtz equation for the complex field amplitude \( P_0(r, z, T) \) formed at point \( (r, z) \) by a tonal source with frequency \( \omega \) or for the spectral component of a wideband signal at this frequency has the form:

\[
P_{\omega}(r, z) = \sum_{m} \frac{C_0}{\sqrt{rq_m}} \psi_m(z) \exp(iq_m r)
\]  

where \( \psi_m(z) \) and \( q_m \) are the eigenfunctions and eigenvalues of the Sturm-Liouville problem, \( \rho \) is the water density, \( k_b = \omega / c_b \) is wavenumber of bottom. For a point isotropic source with power \( W_s \) located at depth \( z_s \) we have

\[
C_0 = \sqrt{\rho_s c_s W_s \exp(i \pi/4) \psi_m(z_s)},
\]

where \( \rho_s \) and \( c_s \) are the water density and sound velocity at depth \( z_s \). In the presence of NIWs, taking into account the smallness of parameter \( |\mu| \sim 0.01 \), the sound field is:

\[
P_{\omega}(r, z, T) = \sum_{m} \frac{C_m(r, T)}{\sqrt{rq_m}} \psi_m(z) \exp(iq_m r),
\]

where for coefficients \( C_m \) we obtain a system of differential equations:

\[
\frac{dC_m}{dr} = i \sum_n V_{mn} C_n \exp(-i \Delta q_{mn} r),
\]

\[
V_{mn}(r, T) = \frac{k^2}{2 \sqrt{q_m q_n}} \int_0^H \mu(r - \nu T, z) \psi_m(z) \psi_n(z) dz \quad \text{and} \quad \Delta q_{mn} = q_m - q_n.
\]

To find the modal coefficients \( C_m(r, T) \) at the receiver, we will consider that NIWs are fully located on the track. Then on the \( r \) axis, three regions are distinguished: (1) the region before NIWs \( r < R \), (2) the region containing NIWs (mode interaction region) \( R < r < R + L \) and (3) the region after NIWs \( r > R + L \). It can easily be seen that in regions (1) and (3), the modal coefficients do not depend on \( r \) and are equal to, respectively, \( C_0 \) and \( C_m(R + L, T) \). Therefore, at the receiver located in region (3), the modal coefficient values are \( C_m(r, T) = C_m(R + L, T) \) and to find them, it is necessary to integrate system (4) over region (2) with the initial condition on the left boundary, \( C_0 \).

We obtain

\[
C_m(r, T) = C_m(R + L, T) = \sum_n S_{mn}(L) C_0 \exp(-i \Delta q_{mn} R),
\]

and

\[
P_{\omega}(r, z, T) = \sum_{m,n} P_{mn} \exp(-i \Delta q_{mn} \nu T),
\]

\[
P_{mn}(r, z) = \frac{C_0}{\sqrt{q_m q_n}} S_{mn}(L) \psi_m(z) \exp(iq_m r).
\]

From formulas (6-7) it is clear that quantity \( P_{\omega}(r, z, T) \) changes with time in accordance
with the set of harmonics $\exp(-i\Delta \Omega_{mn} T)$; i.e., it contains the characteristic frequencies $\Omega_{mn} = v |\Delta q_{mn}|$. (8) 

for typical conditions the scales of interference beating $\Lambda_{mn} = 2\pi/|\Delta q_{mn}|$ have an order of 600-800 m, and the characteristic frequencies $F_{mn} = (2\pi)^{-1} \Omega_{mn} = v/\Lambda_{mn} \sim 4-6$ c/h.

Intensity of the sound field has the following form

$$I_\omega(r,z,T) = \frac{1}{2} \left[ P_{\omega}(r,z,T) \right]^2 = \frac{1}{2} \rho c \sum P_{mn} P_{kl}^* \exp(-i\Omega_{mkl} T)$$

Under real conditions, observation is limited to a finite time interval $T_0 < T < T_0 + \Delta T$ and spectrum of intensity is:

$$G_I(\Omega, \omega) = \left[ \int_{T_0}^{T_0+\Delta T} \left[ I_\omega(r,z,T) - \bar{I}_\omega(r,z) \right] \exp(i\Omega T) dT \right]$$

where $\bar{I}_\omega(r,z) = \frac{1}{\Delta T} \int_{T_0}^{T_0+\Delta T} I_\omega(r,z,T) dT$ is the mean intensity.

if we consider the sound frequency $\omega$ as an independent variable, then the maxima in the fluctuation spectrum in $(\Omega, \omega)$ - the fluctuation rate and sound frequency - are concentrated along lines determined by the dispersion dependences

$$\Omega = \Omega_{mn}(\omega) = v[q_m(\omega) - q_n(\omega)],$$

which depend on the parameters of the unperturbed waveguide. The linewidth along coordinate $\Omega$ is equal to $\Delta \Omega \approx 2\pi/\Delta T$. The distribution of the amplitude of fluctuations along lines (10) without allowing for interference effects in the line-overlap regions depends on the sound frequency and character of perturbations and is determined by the following formula:

$$A_{mn}(\omega) = \Delta T |P_{mn}| = \sqrt{\frac{\Delta T}{rq_{mn}}} |c_0^0 S_{mn}(L) v_m(z)|.$$ (11)

The amplitude of intensity fluctuations at frequencies $\Omega_{mkl}$ are proportional to $|P_{mn} P_{kl}^*|$. 

Due to the smallness of field perturbations caused by NIWs, we have $|P_{mn}| << |P_{mn}|$, since matrix $S$, which is determined by mode interaction, and differs in significantly from the identity matrix $|S_{mn}| << |S_{mn}| \approx 1$. In this connection, in the spectrum of intensity fluctuations, the components for $m = n$ or for $k = l$ are dominant, which corresponds to the same characteristic frequencies $\Omega_{mn}$ as in the complex amplitude spectrum.

Thus, it is possible to consider that the spectra $G_I$ contain identical sets of discrete frequencies $\Omega_{mn}$ determined by the velocity of NIWs and the scales on intermode interference beating in direction from the source to receiver.

3. NUMERICAL MODELING OF INTENSITY FLUCTUATIONS

For numerical modelling the following parameters of waveguide were taken (Fig.1):

$H=88$m, $r=10$ km; $c_1=1484$ m/s, $c_2=1534$ m/s, $z_0=10$ m, $z_1=35$ m, $\varepsilon=(c_2-c_1)/(z_1-z_0)$; $c_b=1700$ m/s, $\rho_b=1.8$ g/cm$^3$, $q_0=0.3$ dB/λ, $R=R_0 + vT$, $R_0=3080$ m, $v= v_s / \sin \alpha$, $v_s=0.8$ m/s, $\alpha=90^\circ$. Sound speed profile in the presence of ISs,
where \( u = x - R \), and
\[
\eta(u) = a \sech \left( \frac{u}{L} \right).
\]

Here \( a = 10 \) m, \( L = L_s / \sin \alpha \), and \( L_s = 95 \) m.

Figure 2a presents, based on calculations of eigenvalues \( q_m(\omega) \) for the given waveguide, the dependences of the intermode beat scales \( \Delta = \Lambda_{mn}(\omega) = 2\pi / \Delta q_{mn}(\omega) \) on sound frequency \( \omega = (2\pi)^{-1} f \). Figure 2b depicts the dispersion lines for \( v = 0.8 \) m/s:
\[
F = F_{mn}(\omega) = (2\pi)^{-1} v \Delta q_{mn}(\omega).
\]

Fluctuations in spectra \( G_i \) can be observed only in the vicinity of the given dispersion lines (see Fig. 3a), which are constructed based for the unperturbed waveguide. The distribution of fluctuation amplitudes along these lines (see Fig. 3b) are determined by the parameters of an unperturbed channel, radiated mode composition and the character of the perturbation.

**Fig. 2. Dispersion curves a) \( \Lambda_{mn}(\omega) \) and b) \( \Omega_{mn}(\omega) \). Vertical lines denote high frequency (ray) limit for predominating frequency (in give case about 3.7 cph).**

**Fig. 3. Spectrum of intensity (a) and distribution of global maximums (b). Depth of the source and receivers are shown. Gradation in dB scale is shown in the right.**

Distribution of spectral intensity over depth for sound frequencies 300 and 500 Hz calculated using PE approximation are shown in the Fig.4 (color diagrams). We can see modal vertical structure, corresponding to numbers of modes, corresponding to the
most significant coupling due to soliton’s motion (7-8\textsuperscript{th} for 300 Hz, and 11-12 for 500 H) in color diagrams. In the right side spectra of intensity fluctuations averaged over depth are shown. We can clearly see harmonics of predominating frequency taking place due to finite time interval for Fourier transform (9).

![Fig.4 Spectra of intensity fluctuations for 300 Hz and 500 Hz of the sound frequency. Depth dependence in the left side and averaged over depth in the right side are shown.](image)

4. CONCLUSION

Thus, as some perturbation moves along an acoustic track, sound field fluctuations can be observed, resulting from mode coupling. Note that in the considered formulation of the problem (the unperturbed waveguide is horizontally homogeneous; NIWs unchanged in shape move along the track, not passing through the source or receiver), the adiabatic component of field fluctuations is completely absent. The spectrum of fluctuations caused by mode interaction has a quasi-discrete character, where the frequency values, which determine the positions of "spectral lines," are proportional to the motion velocity of NIWs and inversely proportional to the scale of interference beats of the interacting modes. The amplitudes of the corresponding spectral components are determined by the form of inhomogeneity, the mode composition of the emitted signal, the sound frequency, and the parameters of the unperturbed waveguide. In situation with sharp thermocline, singular-NIW-type inhomogeneity), the fluctuation spectrum is determined by the interaction of only neighboring modes; therefore, the region of possible dominant frequency values narrows significantly

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STATISTICAL CHARACTERIZATION OF WIDEBAND CHANNEL
IMPULSE RESPONSE OBSERVATIONS IN SHALLOW WATER

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Abstract: The Kauai Acomms MURI 2011 (KAM11) Experiment was conducted in shallow water (~100 m deep) off the western side of Kauai, Hawaii, in June-July 2011. Wideband channel impulse response transmissions (24 kHz bandwidth LFM chirps and MLSs centered at 23 kHz) were carried out every 2 hours for an extended period of time. These were transmitted from a moored 8-element source array with 7.5 m element spacing and carried out in a round-robin fashion for 60 s per source element. A pair of 16-element receive arrays with 3.75 m element spacing were moored at ranges of 3 and 7 km. The fixed source, fixed receiving array geometry enabled observing environmentally-induced fluctuations in the channel impulse response. Selected examples of the temporal variability of the wideband channel impulse response for various source-receiver pairs are shown. Individual eigenray paths are identified and their fluctuation characteristics quantified including temporal correlation scales as well as the relatedness of fluctuations between pairs of paths. In addition, the relatedness of path fluctuations between a pair of elements also is shown. These path and cross-path characteristics then can be used for tapped delay line model channel simulation purposes.

Keywords: wideband, channel impulse response, fluctuations, shallow water, acoustic communication
1. INTRODUCTION

The Kauai Acomms MURI 2011 (KAM11) Experiment was conducted in shallow water (~100 m deep) off the western side of Kauai, Hawaii, over the period 23 June – 12 July 2011 [1]. The objective of KAM11 was to obtain acoustic and environmental data appropriate for studying the coupling of oceanography, acoustics, and underwater communications. Of specific interest was to collect acoustic and environmental data that will relate the impact of a fluctuating oceanographic environment and source/receiver motion to fluctuations in the impulse response of the acoustic channel between multiple sources and receivers and ultimately to the design and performance characterization of acoustic digital data communication systems in shallow water.

The focus of KAM11 was on fluctuations over scales of a tenth of a second to a few tens of seconds that directly affect the reception of a data packet and packet-to-packet variability. While the complete set of data collected is described in [1], the focus here is on a two-day period devoted to wideband transmissions. Specifically, wideband channel impulse response transmissions (24 kHz bandwidth LFM chirps and MLSs centered at 23 kHz) were carried out every 2 hours for an extended period of time. These were transmitted from a moored 8-element source array with 7.5 m element spacing and carried out in a round-robin fashion for 60 s per source element. A pair of 16-element receive arrays with 3.75 m element spacing were moored at ranges of 3 and 7 km from the source array. The fixed source, fixed receiving array geometry enabled observing environmentally-induced fluctuations in the channel impulse response.

2. ENVIRONMENTAL MEASUREMENTS

An example of the dynamic water column environment observed during KAM11 on JD 184 is shown in Fig. 1. The temperature profiles were recorded on a thermistor string deployed between the source array and receive array at 3 km range. The mixed layer depth changes from as little as 20 m to as much as 60 m or more over the course of 24 hours. Similarly, the sea surface exhibited a highly dynamic wave field driven by a daily wind speed pattern that varied from calm conditions of ~2 m/s to whitecap coverage at ~14 m/s [1]. The data discussed here is from 0351 UTC.

3. MULTIPATH STRUCTURE

The multipath structure at 0351 UTC predicted using the Bellhop ray tracing code from Source 1 to Receivers 1 and 3 is shown in Fig. 2. The sound speed profile has a mixed layer depth of ~35 m then a negative gradient thermocline extending to the bottom. As a result, higher-angle ray paths interact with the sea surface while lower-angle ray paths are ducted as near-seafloor refracting rays. The single surface reflecting paths arrive first at the array while the deep refracting rays arrive later.
4. CHANNEL IMPULSE RESPONSE AND FLUCTUATIONS

The wideband (10-34 kHz LFM chirps, 48 ms in duration repeated every 96 ms) time-evolving (over 60 s) channel impulse response (CIR) is shown in Fig. 3 for the deepest source (Source 1) to two deep receiving array elements (Receivers 1 and 3) at a range of 3 km from the source array. The first few arrivals (44-47 ms) correspond to single surface-reflecting paths (the bulk travel time has been removed from these CIR plots). In an effort to remove any residual mooring motion, Fig. 4 is from resampled time series data using as a reference the deep refracting paths arriving at 48.43 ms and 48.21 ms, respectively. The relatedness of fluctuations between pairs of paths is summarized by the path-path covariance matrices [3]. These are shown in Fig. 5 for the individual receivers and in Fig. 6 for the cross-receiver case.

5. SUMMARY

The KAM11 experiment provided an opportunity to observe the time-evolving channel impulse response from a fixed array of sources to a fixed array of receivers over a wide range of sound speed and sea surface conditions. For a two-day period, wideband channel probing transmissions were made along with the transmission of a variety of communication waveforms. The time-evolving channel impulse responses from a deep source to a pair of deep receiving elements were shown along with the relatedness of fluctuations between pairs of paths summarized by the path-path covariance structure for each receiver and cross-covariance between receivers. These covariance matrices then can be used for tapped delay line model channel simulation purposes [4].

6. ACKNOWLEDGEMENTS

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Fig. 1: Temperature profiles recorded on a thermistor string deployed between the source array and 3 km receive array along with wind speed/direction on 3 July (JD 184).

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Session 36

Acoustic Inversions
GEOACOUSTIC INVERSION USING PILE DRIVING PULSE 
AND SURFACE SHIP NOISE OF OPPORTUNITY BASED ON 
SINGLE VECTOR SENSOR

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Abstract: A geoacoustic inversion method using pile driving pulse and surface ship 
noise of opportunity based on single vector sensor was presented here to estimate 
geoacoustic parameters. Firstly, the propagation between a couple of underwater 
source and receiver in the case of a real waveguide was numerically studied via a 
predefined cost function to analyze the geoacoustic parameters’ sensitivity to the 
cutoff frequency and interference range. Then, the cutoff frequency and interference 
range were estimated on the real data using transformation like short-time Fourier 
transform for the pile driving pulse and surface ship noise respectively. And finally the 
best-fit solutions involving sediment sound speed and its thickness and basement 
sound speed etc were found using the defined cost function about the cutoff frequency 
and interference range. The idea was effectively validated during the experiment that 
took place in June of 2013 in South China Sea nearly 100m depth. [Work supported 
by the National 863 Project (No. 2011AA090502) and Defense Industrial Technology 
Development Program (B2420132004).]

Keywords: Geoacoustic inversion; vector sensor; cutoff frequency; interference range
1 INTRODUCTION

Ocean-bottom properties have a major influence in the context of shallow water acoustics especially in the very low frequency. So geoacoustic inversion rather than in-situ measurement for such properties has received considerable attention in recent years for further applications involving target detection and localization etc [1-2]. It can be summarized from three aspects: (1) measurement manners; (2) utilization manners; (3) measurement platforms for sound field information. There is a trend to utilize vector sensor to acquire the interference structure of sound field in passive mode to fulfill the ocean-bottom properties inversion [1-6].

In this paper, a geoacoustic inversion method using pile driving pulse and surface ship noise of opportunity based on the cutoff frequency and interference range recorded and post-analyzed by vector sensor was presented to estimate geoacoustic parameters involving sediment sound speed and its thickness and basement sound speed etc. In the first section, it introduces the advancement about the geoacoustic inversion technology; In section two, the cost function will be constructed and analyzed for its sensitivity for different ocean bottom parameters; The third section will present the sea trial results and its validation; It is summarized in the final section and gives some suggestions for further work.

2 COST FUNCTION CONSTRUCTION AND SENSITIVITY ANALYSIS

2.1 Inversion cost function construction

Due to the influences of water parameters including its density $\rho_w$, sound speed $c_w(z)$, and depth $H_w$, sediment parameters including its density $\rho_s$, sound speed $c_s$, and thickness $H_s$ etc, and basement parameters involving its density $\rho_b$, sound speed $c_b$ etc, the duct can only have less than $n$ modes when the source frequency is lower than some order’s cutoff frequency. In addition, the interference range of the sound field in some frequency (like only holding two orders normal mode) is mainly related to its primary horizontal wavenumber $k_m$ and $k_n$ [7]. If the cutoff frequency of each order normal mode and the interference range of some frequency can be determined by post-processing of the received signal by vector sensor, it is likely to obtain the ocean bottom parameters via geoacoustic inversion method.

Utilizing the cutoff frequency and interference range of the waveguide, the following cost function is constructed:
\[ CF(c_s, \rho_s, H_s, c_b, \rho_b) = \frac{1}{\left( \sum_{m=1}^{M} (f_{c_t}(m) - f_{c_r}(m))^2 \right) \left( \sum_{n=1}^{N} (R_{t}(n) - R_{r}(n))^2 \right)} \]  

(1)

where, \( f_{c_t} \) and \( f_{c_r} \) are the theoretic prediction and real measurement for the cutoff frequency respectively; \( R_{t} \) and \( R_{r} \) the theoretic prediction and real measurement for the interference range respectively. If the theoretic prediction and real measurement are perfectly matched, the cost function will approach to infinity.

### 2.2 Sensitivity analysis for inversion cost function

Constructing the following simulation scenario: \( c_w(z) = 1520 \text{m/s}, \rho_w = 1024 \text{kg/m}^3 \), \( H_w = 85 \text{m} \), \( c_s = 1650 \text{m/s}, \rho_s = 1600 \text{kg/m}^3 \), \( h_s = 39 \text{m} \), \( c_b = 1810 \text{m/s} \), \( \rho_b = 2025 \text{kg/m}^3 \), and the absorptions are omitted for each medium. The cutoff frequency for the second and third order normal mode can calculated as about 21Hz and 30Hz using Kraken routine, and the interference range for 25Hz is 610m.

The sensitivity analysis results for sediment parameters including its density, sound speed and thickness are depicted in Fig.1. And Fig.2 gives the sensitivity analysis results for basement parameters including its density and sound speed. As can be seen from it, the predefined cost function is sensitive to the mentioned parameters above with a narrow lobe.

**Fig.1: Sensitivity analysis results for sediment parameters**

**Fig.2: Sensitivity analysis results for basement parameters**
3 SEA TRIAL RESULTS AND VALIDATION

3.1 Trial data analysis

For the pile driving pulse, the short-time Fourier transformation (STFT) is used to obtain the cutoff frequency of the trial duct. The STFT formula about the sound pressure and the vertical particle velocity are listed as follows:

\[ P(r, z, f) = \text{STFT}[p(r, z, t)] \] (2)

\[ V_z(r, z, f) = \text{STFT}[v_z(r, z, t)] \] (3)

As illustrated in Fig.3~Fig.5, they give the pile driving pulse, STFT analysis result for sound pressure and vertical particle velocity respectively. Every graph has its abscissa of time, and the vertical coordinates are Volts and frequency from Fig.3 to Fig.5. It can be inferred that the first and second order normal mode is 6Hz and 18Hz, which will be used in the following inversion.

For the surface ship noise of opportunity, “\( f - k_r \)” expression is introduced to describe the propagation mode in the waveguide, which is accomplished by taking the Fourier transformation in range dimension (\( r \)) after the time-frequency analysis for the surface ship radiated noise. In the condition of \( r(t) \) (the range between the surface ship and vector sensor) known beforehand, the time-frequency analysis for the surface ship radiated noise can be expressed as [3]:

\[ P(r, f) = S(f) \left( \sum_n A_n^2 + 2 \sum_{n,m} A_n A_m \cos[\Delta k_{mn} r(t)] \right) \] (4)

Taking the Fourier transformation in range dimension, it can get:

\[ P(k_r, f) = 2S(f) \sum_{n,m} A_n A_m \delta(k - \Delta k_{mn}) \] (5)

For Eq.(5), it is needed to eliminate the impact of stem \( P(0,0) \). And it is easy to determine the interference range via Eq.(5):
\[ R = \frac{2\pi}{\Delta k_{mn}} \] 

Fig.6 and Fig.7 give the LOFAR spectrum and \( f-k \) analysis for surface ship noise. According to Eq.(5), the interference range for 25Hz and 32Hz are 848.4m and 905.1m respectively. And the interference range for 32Hz is consistent with the real data (982.2m) in Fig.8.

3.2 Inversion results and validation

According to the defined cost function in Eq.(1) and the estimated cutoff frequency and interference range in section 3.1, the inverted results is shown in Fig.10. And the inverted sediment parameters including its density \( \rho_s \), sound speed \( c_s \), and thickness \( H_s \) are 1580kg/m^3, 1580m/s, and 46m. The basemen parameters involving its density \( \rho_b \) and sound speed \( c_b \) are 2100kg/m^3 and 1810m/s.

In order to validate the inverted results, a sound propagation trial of variable source depth was designed in June of 2013 in South China Sea nearly 100m depth. The UW350 was deployed from 10m water depth to 60m with a depth interval of 10m in range of not less than 800m, and the transmitted signal was 30Hz CW waveform. The trial result is shown in Fig.9. As can be seen from it, the received signal spectrum level and the phase difference between \( p \) and \( v_z \) for 30Hz had the theoretic prediction and real measurement consistent with each other, which indicated that geoaoustic inversion method using pile driving pulse and surface ship noise of opportunity based on single vector sensor was effective and the inverted parameters were reasonable.
4 CONCLUSION

A geoacoustic inversion method based on single vector sensor was presented here to estimate ocean-bottom properties, which made use of the cutoff frequency and interference range of the waveguide to construct the inversion cost function. In one hand, the cutoff frequency was obtained by taking STFT for the pile driving pulse; On the other hand, the interference range was determined via introducing “$f - k_r$” expression for the surface ship noise of opportunity. Sea trial data analysis indicated that this method had its effectiveness and reasonability, and the inverted parameters had been validated through designed sound propagation trial. Moreover, there is needed to utilizing the comprehensive information of the vector field to use only the surface ship radiated noise.

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Fig. 10: Inversion results
INFERRING OCEAN TEMPERATURE VARIATIONS FROM SHIPPING NOISE

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Abstract: Acoustic passive methods to infer properties of the oceanic medium have been subject of progressive emphasis in order to obtain low cost, environmental friendly, long time characterization of the ocean. To this end a first step consists in the estimation of the frequency response of the medium or its time domain counterpart, the impulse response. In this work we consider distant ship noise as an opportunity source characterized by a few low frequency discrete tones. Therefore, the frequency response of a shallow water acoustic channel is estimated at these discrete frequencies, between two vertical line arrays (VLA's). The influence on the estimates of several factors of uncertainty such as range, depth and tilt variations in the VLA's is investigated. For validation purposes, a phase conjugation method is considered. In a preliminary approach towards passive ocean acoustic tomography implementation the estimates obtained by the proposed method are applied in a matched field framework to track sound speed profile variations. Simulations are conducted based on experimental setup and environmental parameters gathered during the MREA07 sea trial that took place in the Tyrrhenian Sea, near Elba Island in May 2007. The results show that, although shadowed, the obtained frequency response estimates allows to obtain a permanent focus and enables the tracking of sound speed profile variations in the water column.

Keywords: Passive inverse methods, coherent noise processing, ocean acoustic passive tomography
1. INTRODUCTION

In underwater acoustic applications (acoustic communications, source localization, environmental monitoring) noise is usually considered a nuisance factor, therefore, several methods have been developed to minimize its impact. However, the underwater environment is naturally noisy and in last years passive methods have been subject of growing interest (see [1] and references herein). These methods use ambient noise, either natural or anthropogenic, instead of active sources. Therefore they are considered to be low cost and environmentally friendly methods and are able to be applied for long periods of time. Predominant acoustic noise in the ocean is due to shipping, which is characterized by a few low discrete frequency tones superimposed on a diffuse background pedestal, travelling over long distances and carrying water column structure information. Nowadays there is a vast number of near shore maritime routes with high levels of traffic, where it would be simple to install receiver arrays, thereby enabling low cost spatial-temporal monitoring of oceanographic processes.

In this work we consider long range ship noise as opportunity sources for tracking sound speed/temperature perturbations. The waveguide frequency response estimation is developed based on cross-correlation methods, using a normal mode approach [2, 3]. The MREA07 geometry is used to perform simulations, where two VLAs at 4 km distance are receiving the ship noise. The method is validated with a phase conjugation procedure and it is shown that the inclusion of uncertainty factors in the VLAs position can still be accounted for. Finally it is shown that the tracking of sound speed/temperature variations is possible.

2. MODEL DEVELOPMENT

We consider the ship noise as a monotonic source of fixed frequency $\omega_0$ characterized by strength (amplitude) $A$ and a phase $\varphi$, wide-sense stationary and ergodic stochastic processes. The ship is at location $S$, two receivers are at locations $A$ and $B$ and the propagation channels are considered as linear systems. Assuming that the frequency response at frequency $\omega_0$, between the source and the receivers are respectively $H_{SA}(\omega_0)e^{j\varphi_{SA}(\omega_0)}$ and $H_{BA}(\omega_0)e^{j\varphi_{SB}(\omega_0)}$, where $H_{SX}$ is the module and $\varphi_{SX}$ is the phase, considering the receiver at location $X$. Therefore, the received signal at points $A$ and $B$ can be written as

$$y_A(\omega_0, t) = A(t)e^{j(\omega_0 t + \varphi(t))}H_{SA}(\omega_0)e^{j\varphi_{SA}(\omega_0)} + n_A(t),$$

$$y_B(\omega_0, t) = A(t)e^{j(\omega_0 t + \varphi(t))}H_{SB}(\omega_0)e^{j\varphi_{SB}(\omega_0)} + n_B(t),$$

where $n_A(t)$ and $n_B(t)$ are uncorrelated additive zero mean noise components, uncorrelated with the signal. One can write the cross-correlation function at 0 lag (i.e. cross power) between the receiver $A$ and receiver $B$ as

$$r_{AB} = E[y_A(\omega_0, t)y_B^*(\omega_0, t)],$$

where $E[.]$ represents the expectation operator. Assuming a deterministic behavior of the frequency responses this cross-correlation function between two receivers can be written as

$$r_{AB} = E[A^2(t)]H_{SA}(\omega_0)H_{SB}(\omega_0)e^{j(\varphi_{SA} - \varphi_{SB})} + E[n_A(t)n_B^*(t)],$$
where \( E[A^2(t)] \) represents the variance of \( A(t) \) and is assumed constant. As it can be observed, the cross-power does not depend on the initial phase \( \phi \). The term \( E[n_A(t)n_B(t)] \) represents the cross-power of the noise and under the assumptions made it will vanish.

The frequency response between the source and the receiver using the normal mode long range approximation can be written as \([2, 3]\)

\[
H_{\omega_0}(R_{SA}, z_S, z_A) \propto \sum_{n=1}^{N} U_n(z_S) U_n(z_A) \frac{e^{ik_nR_{SA}}}{\sqrt{k_nR_{SA}}},
\]

where \( R_{SA} \) is the range between S and A, \( S \) is the source and \( A \) is a receiver, \( z_X \) represents the depth associated with \( X \), \( N \) is the number of modal functions \( U_n \) and \( k_n \) is the corresponding horizontal wavenumber. For a matter of simplicity we will indicate the frequency response as \( H(S, A) \). Using a similar approach as in \([2]\) and assuming the reciprocity of the medium, an estimate \( \tilde{H}(S, X_k) \) for the frequency response between the source \( S \) and the receiver array \( X \), where \( X_k \) is the k-th receiver of array \( X \), \( X \) being \( A \) or \( B \), were obtained as

\[
\tilde{H}_I(S, A), \text{ where } \tilde{H}_I(S, A_k) = \sum_i H(B_i, A_k)H(B_i, S).
\]

\[
\tilde{H}_O(S, B), \text{ where } \tilde{H}_O(S, B_k) = \sum_i H(A_i, S)H^*(A_i, B_k),
\]

the subscript \( I \) means that Eq. (6) was obtained using the receivers of the inner array \( B \), while the subscript \( O \) means that Eq. (7) was obtained using the receivers of the outer array \( A \). Assuming the normal mode approximation the cross-correlation between the receivers in arrays \( A \) and \( B \) can be written in a matrix form as

\[
H^s(A, B), \text{ with } H_{k,I}^s \propto \sum_n \frac{U^2_2(S)}{\sqrt{k_n}} \tilde{H}(A_k, B_I),
\]

where \( H_{k,I}^s \) can be seen as proportional to the term \( H_{SA}(\omega_0)H_{SB}(\omega_0)e^{j(\phi_{SA}-\phi_{SB})} \) in Eq. (4).

2.1. Model validation

In this section we consider two approaches for validation purposes of the estimates obtained above. The first is a phase conjugation procedure for source localization; the second is a matched field based method for sound speed estimation.

2.1.1. Phase Conjugation

Phase conjugation, or its time domain counterpart time reversal, is a well known method of sound refocusing that had been spread in time and space by propagation through the ocean. In this work we compare a reference field, given by a normal mode propagation model and the focused field obtained with the estimates developed above according to

\[
PC(z, \omega) = \sum_{A_i} H^*(A_i, B_0) H(A_i, z).
\]

\[
\bar{PC}(z, \omega) = \sum_{A_i} H^s(A_i, B_0) H(A_i, z),
\]

where \( PC \) indicates the reference focused field obtained with the frequency responses given by the propagation model; \( \bar{PC} \) is the estimated focused field, obtained via the estimated frequency response \( H^s \) and \( z \) represents the depth at, or near to, the focal point.
2.1.2. Matched field

Matched field processing methods have been used for tomography purposes [4]. In this work we apply this technique to estimate perturbations in the sound speed profile, which occur usually due to the variations of day cycle. One of the most robust methods in matched field processing is the Bartlett processor that can be written as

\[ P_{Bart}(\phi) = \frac{1}{N_B N_\omega} \sum_\omega \sum_i \left| (H_0^s(A, B_i, \omega, \phi)) H_0^i(A, B_i, \omega, \phi_0) \right|^2, \]

where \( P_{Bart} \) is the Bartlett power, \( \phi \) represents the sound speed profile and \( \phi_0 \) is the mean sound speed profile, \( N_B \) is the number of receivers in VLA B, \( N_\omega \) is the number of frequencies, the subscript 0 stands for norm2 normalization of \( H^s \) and \( H_0 \), \( B_i \) is the \( i \)th receiver of VLA B, \( \omega \) is the frequency and the superscript \( H \) stands for conjugate transpose operation. \( P_{Bart} \) is a broadband processor incoherent in both frequency and space.

3. SIMULATIONS

In this section a simulation scenario is considered, based on the MREA07 sea trial experiment [5] that took place near the Elba Island, in the Tyrrhenian Sea. The geometry is depicted in Fig. 1 (left). The environment is range independent with a 112m depth water column above a sediment half-space. Fig. 1 shows on the right hand side the sound speed profiles collected during the MREA07 sea trial (dashed) and the mean profile (solid). The geometry setup reflects the positioning of the two VLAs used in the sea trial. The distance between the VLAs is 4 km, the VLA B distance to the ship is initially 6 km and then slowly decreasing to 4 km. The VLA A has 16 hydrophones, equally spaced from 6 to 66 m depth. The VLA B has 8 hydrophones, two at 9 and 14m, and the other six are equally spaced from 55 to 79 m depth. The numerical simulations were obtained with the normal mode model KRAKEN [6]. The estimated frequency response in the 100-300 Hz band, between the two VLAs obtained in Eq. (8) is assessed by means of the phase conjugation method described in the previous section. Fig. 2 shows the space-time ambiguity surface of the focused field at the 2nd hydrophone of the VLA B where one compares the reference focused field (a), obtained via Eq. (9), with the noise cross-correlation estimated focused field using Eq. (10) (b), revealing a good agreement in ambiguity surfaces. It is clear in both plots that the peak is at true depth (14m). Temporally
the focus is also well compressed at zero lag, although as it would be expected, with higher levels of ambiguity in the noise cross-correlation estimated focused field.

![Fig. 2: Comparison of the spatial-temporal representation of the focused fields on the 2nd hydrophone of VLA B: (a) - focused field, using Eq. (9); (b) – noise cross-correlation estimated focused field using Eq.(10); (c) – mean noise cross-correlation estimated focused field affected by random array tilt perturbation.](image)

Next we consider the influence on the estimated frequency response of uncertainty factors in the VLAs position: distance between the arrays, depth and tilt, individually taken into account. These uncertainties were modeled using Gaussian random perturbations to simulate GPS uncertainty and surface waves and Uniform random perturbations to simulate ocean current effects. Standard deviations of 5 m, 50 cm and variance of 3 deg, were used for array range, sensor depth and array tilt respectively. For each uncertainty factor a set of 200 realizations was performed where for each realization the noise cross-correlation estimated field was obtained. Afterwards the mean noise cross-correlation estimated field was used with the phase conjugation method to assess the influence of the parameters perturbation. Fig. 2 (c) shows the influence of tilt perturbations which revealed overall the same behavior as when the perturbations were not considered. This behavior was also noticed for the other uncertainty factors. The model used to study the uncertainty factors in the VLAs position will be further investigated in future work.

![Fig. 3: Left–spectrogram of a radiated ship noise up to 400 Hz band; Right–Bartlett processor in frequency responses comparison of Model/Model (blue) and Estimates/Model (red).](image)

According to the matched field based approach we investigated the ability of the estimated frequency responses between the VLAs, Eq. (8), to track sound speed profiles variation. For this purpose, the collected sound speed profiles in the MREA07 sea trial represented in Fig. 1 on the right (dashed) were used. Fig. 3 on the left presents the spectrogram of the radiated noise of a research vessel, where a set of few low frequency tones is clearly observable up to 400 Hz. Therefore, in the simulations, a set of 11 equally spaced discrete frequencies were
considered in the 100-300 Hz band. Fig. 3 (right) depicts the behavior of the Bartlett processor, obtained with Eq. (11), using only the frequency responses given by the propagation model (blue) and using the noise cross-correlation estimates (red). The x-axis represents the normalized sound speed perturbation index $\alpha$, $-1 \leq \alpha \leq 1$. The used sound speed $\phi(z) = \phi_0(z) + \alpha(\phi_{\text{min}}(z) - \phi_0(z))$, for negative $\alpha$, and $\phi(z) = \phi_0(z) + \alpha(\phi_{\text{max}}(z) - \phi_0(z))$, for positive $\alpha$, where $\phi_0$ is the mean sound speed profile represented in Fig. 1 (solid), $\phi_{\text{min}}$ and $\phi_{\text{max}}$ are respectively the minimum and maximum collected values of sound speed profiles. The trend of the processor reveals that the estimated frequency responses are able to track variations in the sound speed profiles, although the peak is lower and broader.

4. CONCLUSIONS

In this paper we consider a passive method for the estimation of the underwater acoustic channel frequency response, using a normal mode approximation, where the opportunity source is the ship noise at long range. It is shown that it is possible to characterize the acoustic channel between two arrays at discrete frequencies, typically produced by distant ships in a realistic coastal shallow water scenario. Moreover, it was demonstrated through simulations that the frequency response estimates of the acoustic channel can be used in a matched field inversion procedure to track sound speed/temperature perturbations between the two arrays. This work is a contribution for the usage of noise cross-correlation methods to estimate the sound speed perturbations in shallow coastal water.

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MARINE MAMMAL’S DIRECTIVITY IN GEOACOUSTIC INVERSION SCHEME

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Abstract: Gervaise & al 2011 and Barazzutti & al 2013 described the general structure of a scheme to estimate the nature of superficial sediment in shallow waters using marine mammal’s whistles and a single receiver. The multipath structure of calls given by a spectrogram is used to estimate the source characteristics and the superficial sea bottom features. A field application of this method was presented in [11] using controlled signals similar to marine mammal’s vocalizations in a shallow water environment on a sandy bottom. However, contrary to the source used during that experiment, marine mammals are directive sources and the directivity loss underwent by the multipath must be taken into account in our inversion process. Indeed, the directivity is a function of frequency and emission angle (sound-source azimuth), and impacts each path differently. Thus the bottom path, once corrected from transmission loss, must be corrected from directivity loss before being used to estimate the bottom features. The emission angle can easily be geometrically related to the arrival angle and a specific unknown angle we called attitude (source orientation in space during the emission). However, the directivity patterns of marine mammals are not well studied yet, especially for vocalizations (e.g. directivity model assumption – Au 1993[4], directivity pattern measurement – Au & al 2012[9], etc.) and contrary to other mammals the unknown “attitude” parameter is not that easy to observe (e.g. Dantzker & al 1999). Our communication aims at describing different methods to estimate the “attitude” angle and the directivity loss for marine mammals. Their performances and limits are evaluated using simulated data.

Keywords: Directivity, Inversion, Geoacoustic, Marine Mammals
1. INTRODUCTION AND CONTEXT

Directivity both on land and undersea – In mentioned emission beam pattern studies ([1] to [9]), authors aim at finding satisfying models to understand the emission pattern observations. For nasally emitting bat for instance, sound emission through the nostrils can be approached by two emitters close enough to interfere and act on the beam directivity, with evidences of a relation between the nostrils separation and the emitted wavelength [1]. For orally emitting bat, [2] demonstrate the relation between the mouth radius and the radius of the circular piston model. Nonetheless, they show that, the piston model explains the directivity direction but not the whole beam pattern, especially the ventral side lobe observed for different species of bats [2, 3]. Undersea, [4] assumes also the model of a circular piston on an infinite baffle as the directivity index model for marine mammals. They give values of the radius for Atlantic bottlenose dolphins (Tursiops truncatus) or Belugas (Delphinapterus leucas). [5] measure some false killer whale (Pseudorca crassidens) transmission pattern and reveal its directivity index can be modelled by a planar rectangular transducer. The male sage grouse (Centrocercus Urophasianus) acoustic emission is highly direct and enables the male to attract females with high-intensity signals while showing them all the same its best profile. The beam pattern of male sage grouse whistles beam pattern is asymmetric about the bird’s anterior-posterior axis and presents a null directivity in front of the bird (contrary to the common beam pattern which main axis quite matches the head orientation - [6]).

All these studies use captive animals or, at least, animals with assessable position, attitude (pitch) and yaw. Indeed [6] study free-flying birds but use video records to assess the position and attitude. [5] and [7] use supervised configurations where the source is trained to take a specific position. This implies the [source - receivers] geometry and main axis orientation are known. What is more, the emissions are often stimulated, electrically for bats ([2], [8]), or with trained exercises for marine mammals ([5], [9], [7]).

Context of our work – The directivity pattern is not the purpose of our work but a mean to access the information we need. [10] and [11] describe the general structure of a scheme to estimate the nature of superficial sediment in shallow waters using marine mammal’s whistles and a single receiver. The multipath structure and levels of calls resolved by spectrogram are used to estimate the source characteristics and the superficial sea bottom features. As marine mammals are directive sources, these levels have to be corrected from directivity losses. Moreover, in our situation, we use free-swimming sources so that we do not have direct access to the [source - receiver] geometry nor to the emitted signal. The location of the source is estimated using multipath arrivals. The attitude of the source is missing. Furthermore, we work with a single hydrophone. Thus, for one whistle, we get few emission angles (those from the multipath structures). The visibility on the beam pattern remains therefore partial.

Content – In this paper, we present briefly the inversion scheme. Then we detail the directivity issue for our inversion scheme and we describe the different methods considered and to be considered.
2. DIRECTIVITY

2.1. Observations
Most of the studies presented deal with clicks and pulses and rarely with whistles ([7], [6]).
We perform three sessions of acoustic records in 2009 and 2010 in Bay of Biscay and Ushant area. Autonomous recorders AURAL from Multi Electronic Inc. were moored in shallow water (130 m) at 85 m depth. An exhaustive exploration of these data with a home-made whistles detector indicates that the 2009’s record present nine hours filled with whistles on a total of 52 hours of measurements, where 9 were covered by dolphins (*Delphinus Delphi*) signals.

Fig. 1 - Whistle from Bay of Biscay recording. Box a1: the second path level is higher than the direct one. Box a2: quite null level.

Hundred whistles have been processed using the first step of the inversion scheme: source localization, integrated level of the first paths. From this evaluation, we learnt about our method (functioning domain (for source-receivers optimal configurations), selected signal features) and about the acoustic behaviour of the met dolphins.

Fig. 1 shows a whistle that carries proofs of directivity. In box a1, the second path level cannot be higher than the direct one since it travelled a longer distance. In a2, the presence of nulls for some frequencies reflects the different attenuations according to the frequency in the path emission direction. Table 1 gives statistics that highlight the directivity of the dolphin whistles.

<table>
<thead>
<tr>
<th>Differences between the broadband level of the direct path and the first reflected path (without transmission loss correction)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Mean</td>
</tr>
<tr>
<td>-0.2364</td>
</tr>
</tbody>
</table>

*Table 1 - Some statistics extracted from the 100 whistles study*

2.2. First method to access directivity features: use of the commonly admitted model
The circular piston in an infinite baffle is a simple model for directional acoustic emission.
where $I_0$ is the intensity on the main axis, $a$ the piston radius, $f$ the frequency, $c$ the sound speed, $\theta$ the angle with the normal to the piston surface, $J_1$ the first-order Bessel function. [4] explains that the acoustic projection system of the dolphin can be modeled by an equivalent piston with the same directivity index and the same near-field/far-field transition distance.

In the path level correction step, the directivity model features are learnt from the direct and first surface reflected paths, both not impacted by the bottom features. Both the attitude of the source and the piston radius (which we consider individual-specific) were estimated. We used the observed level differences between the surface and the direct path as observable and a mean least square optimization ([12]). Equations (2) and (3) give respectively the measurements and the estimate used in the optimization step.

\[
\begin{align*}
\text{measure} &= 10 \log_{10} |SL.H_3 + N|^2 - 10 \log_{10} (|SL.H_d + N|^2) \\
\text{estimate} &= 10 \log_{10} \left( \frac{2J_1(p.2\pi f/c \sin(\theta_d))}{p.2\pi f/c \sin(\theta_d)} \right)^2 - 10 \log_{10} \left( \frac{2J_1(p.2\pi f/c \sin(\theta_d))}{p.2\pi f/c \sin(\theta_d)} \right)^2
\end{align*}
\]

Cramer-Rao studies highlight the performances of this model. With both the radius and attitude estimated, bad performances occur in specific predictable directions. Monte-Carlo simulations on synthetic data confirmed this observation ([12]).

As also observed in [4], [9] and [7], even if the circular piston is a rather good model to explain the directivity index of the dolphin transmission beam, it does not well describe the off-axis shape of the transmission beam pattern. We tried to bypass this observation, including an attenuation constant in the formula to take into account both the ambient noise level and the fact that no sharp nulls are encountered in real dolphin signals.

\[
H(\theta) + \text{Cte.} (1 - H(\theta)) \quad \text{with} \quad H(\theta) = \frac{2J_1(p.2\pi f/c \sin(\theta))}{p.2\pi f/c \sin(\theta)}
\]

With Cte a constant evaluated using the noise to signal ratio. This attenuated model showed a better fit to the data but it keeps limitations such as the only 2D view, a symmetry jaw/melon which contradicts the Tursiops beam pattern measurements ([4]).
Fig. 2 shows one result with the piston and attenuated piston models. The piston radii we got were rather high compared to the literature (0.18 m against 0.04 in [4]) and the differences between the model and the observation were too important to use the approximate model to correct the bottom path from directivity losses.

2.3. Second hint: design and estimation at the same time

[13] presents methods to design an arbitrary beam former response. The general solution to wave equation, driving the beamforming, can be decomposed into modes. The solution can be expressed as a sum of modes of spherical harmonics (see equation (5)).

\[
b(r, \theta, \varphi; k) = \sum_{n=0}^{\infty} \sum_{m=-n}^{n} \alpha_{nm}(k) h_n^{(1)}(kr) Y_{nm}(\theta, \varphi)
\]

with \(\alpha_{nm}(k)\) a set of frequency dependent modal coefficient, \(k=2\pi f/c\), \(h_n^{(1)}\) the spherical hankel function of the first kind, \(Y_{nm}\) spherical harmonics operating Legendre polynomials. The analysis equation (projection of (5) on spherical harmonics base) gives the \(\alpha_{nm}(k)\) coefficients for an arbitrary beam pattern. E.g. a piston diagram can be built using 4 modes (for less than 5% of relative error), see Fig. 3.

We dispose, for each whistle, of the level differences between the surface path and the direct path, for different frequencies as shown in equation (2). Using \(n_w\) whistles and \(n_f\) frequency bandwidths centered on \(f_i\) (\(i=1..10\)), we have \(n_w \times n_f\) measurements. In (3), we replace the piston model by the spherical harmonics model (5). We are looking for \(n_w\) attitudes (pitch or elevation angles) and \(n_w\) yaw (or heading) angles. Considering only one common pattern for all the sources, we have to estimate \(\sum_{n=0}^{N}(2n + 1)\) coefficients with \(N\) the number of modes chosen to approximate the pattern. For 4 modes and 100 whistles, we are searching for 225 unknown with 1000 measurements. With some assumptions (e.g. symmetry), the number of coefficients can be reduced.

A global optimization method can be applied to estimate the set of coefficients of the pattern and the angles characterizing the directivity.

3. DISCUSSION AND PERSPECTIVES

The piston model has been evaluated but did not give satisfaction because it creates nulls that are not realistic and even with the best parameters, the bottom level correction will be
compromised by this. The second method is still in progress but seems to offer more flexibility to model asymmetry and to access and correct from directivity. The conditioning study of the model shows how sensitive the model is to the coefficients and angles errors.

REFERENCES


BAYESIAN RECONSTRUCTION OF SEAFLOOR SHAPE FROM SIDE-SCAN SONAR MEASUREMENTS USING A MARKOV RANDOM FIELD

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Abstract: To explore the seafloor, a side-scan sonar emits a directed acoustic signal and then records the returning (reflected) signal intensity as a function of time. The inversion of that process is not unique: multiple shapes may lead to identical measured responses. In this work, we suggest a Bayesian approach to reconstructing the 3D shape of the seafloor from multiple sonar measurements, inspired by the state-of-the-art methods of inverse raytracing that originated in computer vision. The space near the bottom is modelled as a grid of voxels, whose occupancies are represented by random binary variables. Any assignment of occupancies corresponds to some seafloor shape. A global multi-component energy potential describes how well the resulting surface agrees with the sonar data and with the a priori assumptions. Minimization of energy is equivalent to finding the maximum a posteriori (MAP) assignment to this Markov random field (MRF) and is done using the iterated belief propagation (BP) algorithm.

The critical step in this method is to compute messages from “factors” representing the sonar beams to voxels. Naively, its complexity scales exponentially with the number of voxels traversed by a beam. Unlike inverse raytracing, where a pixel value constrains voxels only along a single view ray, a sonar beam involves voxels within a relatively wide cone. Employing dynamic programming techniques and space-filling curves, we were able to develop a practical approximate solution to this problem.

The algorithm is not restricted to side-scan sonar reconstruction and could be applied to medical ultrasound or ultra wide-band (UWB) radar imaging.

Keywords: Bayesian reconstruction, surface estimation, side-scan sonar, inverse raytracing, Markov random field, belief propagation
1. INTRODUCTION

Reconstructing a seafloor 3D shape from side-scan sonar data is difficult since the recorded signal carries only indirect information about the distance from the sensor to bottom. Beam propagation in the water and reflection from the bottom can be modelled relatively accurately, see e.g., [1], [2], and [3] but the inverse problem is to our knowledge not yet solved. A similar problem in computer vision, namely the inverse raytracing, has recently seen an efficient solution [4], [5].

The latter works by Liu and Cooper focus on reconstructing a 3D scene from multiple camera views. To that end, they introduce a grid of voxels spanning a volume of interest. Each voxel corresponds to a binary random variable that describes its occupancy. The relations between voxels are encoded in terms of “factors” linked to those variables. The resulting structure known as a Markov random field (MRF) can be represented as a bi-partite graph. In essence, an MRF describes the joint probability of a simultaneous assignment to all the occupancy variables. A huge monolithic probability function is replaced with a long product of functions each depending only on a few variables, enabling efficient inference methods.

The RayMRF, as Liu and Cooper call it, contains three types of factors. A unit factor is linked to a single variable and describes the a priori occupancy probability for that voxel. Pair factors connect two adjacent voxels and encode, e.g., continuity or smoothness assumptions. The most interesting ray factors are linked to all voxels pierced by a camera view ray corresponding to a single pixel and describe the agreement between the observed pixel value and the given assignment of voxel values.

One practical way to “calibrate” such an MRF, or find the maximum a posteriori (MAP) assignment of voxel variables (or, equivalently, the scene shape), is known as loopy belief propagation (LBP). Each factor iteratively exchanges “messages” with all the linked variables. The messages express the “beliefs” about the occupancy of each voxel based on prior data and the messages from other variables. To compute a message from a factor, one has to consider all assignments to the linked variables and thus solve a local optimization problem. This is not a big problem for, e.g., pair factors where one only has to consider four assignments (for a binary variable), but a ray factor in practice can be linked to hundreds of voxels. The number of assignments, growing exponentially, prohibits a naïve brute-force solution.

RayMRF presents a novel method to compute ray factor messages. It exploits the fact that the ray “energy”, or the assignment likelihood, depends only on the first occupied voxel on a ray. With dynamic programming, the complexity of this step is reduced to linear in the number of linked voxels. In this report we generalize this approach to apply to the (much more challenging) problem of the sonar data inversion.

2. MRF APPLICATION TO SIDE-SCAN SONAR DATA

2.1. Geometry and observables of side-scan sonar beams

Unlike thin camera view rays, a sonar beam has typically a wide and flat power distribution. If by analogy with RayMRF we devise an MRF-based model for sonar-based reconstruction (a “BeamMRF”) with unit, pair, and beam factors, each of the latter will be connected to a
large number of voxel variables (larger than that for a narrow ray). Next, instead of a single RGB value, the observed quantity for a beam is a function \( I(t) \) that represents the reflected acoustic energy that has reached the receiver at the time moment \( t \). In order to relate the time to geometry, we assume isovelocity sound propagation, i.e., a constant and isotropic speed of sound in the water. This translates to straight conical sonar beams. In order to describe the relation between the function \( I(t) \) and the bottom shape it is necessary to discretize the beam geometry. First, we split the wide beam into infinitesimally narrow cones. Second, we sub-divide each cone into individual slices as shown in Figure 1.

![Figure 1: Discretized sonar beam geometry for coinciding source and receiver positions.](image)

The recorded infinitesimal energy \( \Delta R(t_i) \) at some time slice \( t_i \) is a sum of contributions \( \Delta R_s \) from ray cones (indexed by \( s \)) in some set \( S(t_i) \) which encompasses all surface elements that can be reached by the signal (i.e., which are not shaded) and located at a distance determined by the return travel time \( t_i \) (here we neglect multiple reflections):

\[
\Delta R(t_i) = \sum_{s \in S(t_i)} \Delta R_s .
\]

This model can be further described using a two-dimensional grid (Figure 2). Given \( D \) directions inside the beam and \( T \) discernible time slices, the sonar response can be computed based on the occupancies of \( D \cdot T \) voxels:

![Figure 2: Logical voxel structure of a sonar beam](image)

Any assignment of variables on this grid corresponds to some bottom shape. Let us denote the first occupied voxel in direction \( d \) by \( t^*(d) \). That means that all voxels before \( t^*(d) \) (i.e., those with \( t < t^*(d) \)) on the \( d \)-th ray are empty while voxels after \( t^*(d) \) may be either empty or occupied. If we assume that a voxel at \( (t, d) \) reflects back an amount of energy given by \( R(t, d) \) (depending on, e.g., surface normal, its material, and the sonar characteristics), we may compute the energy corresponding to the depth/time slice \( t \) as

\[
\Delta R(t) = \sum_{d=1}^{D} R(t, d) \cdot \delta_{t^*(d)},
\]

where \( \delta \) is the Kronecker delta symbol. Denoting the voxel occupancy variables as \( o(t, d) \in \{0,1\} \), we may further define \( t^*(d) \) as follows:

1 Here we skip the relation between the beam and the world voxels which in the RayMRF model is known as interpolation. It suffices here to say that this problem is solved using the common raytracing methods.
\[ t^*(d) = \begin{cases} 
1, & \text{if } o(1,d) = 1 \\
2, & \text{if } o(1,d) = 0 \land o(2,d) = 1 \\
3, & \text{if } o(1,d) = 0 \land o(2,d) = 0 \land o(3,d) = 1 \\
e tc.
\]

For each beam \( B_t \), the agreement between the observation and the model is given by a quadratic functional ("beam energy", not to be confused with the acoustic energy!):

\[ E_{B_t} = \sum_{t=1}^{T} \left( \Delta R^{\text{estim}}(t) - \Delta R^{\text{observed}}(t) \right)^2 \cdot \varphi(t), \]

with \( \varphi(t) \) being some time-dependent weight that accounts, e.g., for lower signal-to-noise ratio for later readings. The full MRF energy (which can be thought of as the negative logarithm of the joint probability function) contains then the prior terms \( E_u \) and \( E_p \) (unit and pair energies), and the sum of beam energies from the set of observations \( B \). The goal of the reconstruction is to find an assignment \( O \) to voxel occupancies that minimizes the total energy functional:

\[ O_{\text{opt}} = \text{argmin} \left( E_u + E_p + \sum_{B_t \in B} E_{B_t} \right). \]

### 2.2. Messages from beam factors to voxel variables

The details and the justification of the LBP method can be found elsewhere [6]. We also assume that the unit and pair factor updates are performed as in the RayMRF model. Here we focus only on the non-trivial problem of computing the messages from a beam factor to the linked variables. Given the energy functional \( E_{B_t} \), we formally define the needed differential message from the beam factor to voxel at \((t,d)\) as

\[ w(t,d) = M_{f\rightarrow t,d} (o(t,d) = 1) - M_{f\rightarrow t,d} (o(t,d) = 0), \]

\[ M_{f\rightarrow t,d} (o(t,d) = 1) = \min_{o(t',d')} [o(t',d')] \left( E_{B_t}([o(t',d')]) + \sum_{(t',d') \neq (t,d)} M_{td\rightarrow f} (o(t,d)) \right), \]

\[ M_{f\rightarrow t,d} (o(t,d) = 0) = \min_{o(t',d')} [o(t',d')] \left( E_{B_t}([o(t',d')]) + \sum_{(t',d') \neq (t,d)} M_{td\rightarrow f} (o(t,d)) \right). \]

The incoming messages \( M_{td\rightarrow f} (o(t,d)) \) are known, and the minimum in each case is taken over all assignments to variables other than \((t,d)\). The above formula requires thus roughly \( O(2^{T\cdot D}) \) steps, which is too expensive for any reasonable \( T \) and \( D \).

### 2.3. Exact beam front-based solution

Following Liu and Cooper, we notice that the beam energy in fact only depends on \( t^*(d) \). The summations above can therefore be split into three regions (Figure 3). The voxels at \((t^*(d),d)\) belong to region \( A \), those at \((t,d), t > t^*(d)\) to region \( B \), and those at \((t,d), t < t^*(d)\) to region \( C \). Note that in region \( A \) all voxels are necessarily occupied, and in region \( C \) necessarily empty.
Simplifying terms in each region, we may compute the messages with a minimum taken over all assignments to \( t^*(d) = (a_1, ..., a_D) = \bar{a} \) instead of the binary grid:

\[
M_{f \rightarrow d}(o(t, d) = 1) = \min_{\bar{a}[a_d \leq t]} \left( E_B(\{\bar{a}\}) + \sum_{d=1}^{D} m(a_d, d') + \sum_{d=1}^{D} \sum_{t'=a_{d+1}}^{T} \min(0, m(t', d')) - C_T - C_T \right) + \xi
\]

\[
CT_1 := \text{if } t > a_d : \min(0, m(t, d)), \text{else } 0
\]

\[
CT_2 := \text{if } t = a_d : m(t, d), \text{else } 0.
\]

Since an unoccupied voxel at \((t, d)\) can only belong to regions \(C\) or \(B\), we have:

\[
M_{f \rightarrow d}(o(t, d) = 0) = \min_{\bar{a}[a_d \neq t]} \left( E_B(\{\bar{a}\}) + \sum_{d=1}^{D} m(a_d, d') + \sum_{d=1}^{D} \sum_{t'=a_{d+1}}^{T} \min(0, m(t', d')) - C_T \right) + \xi
\]

\[
CT_3 := \text{if } t > a_d : \min(0, m(t, d)), \text{else } 0.
\]

The constant \(\xi\) cancels in the final formula for \(w(t, d)\) and is thus not important. This representation is exact and has complexity \(O(D^T)\). The double sums under minima would naïvely require more steps but the iteration over \(t^*(d)\) can be re-organized so that each sum can be incrementally updated in constant time on each step. The same applies to beam energies of each configuration. Moreover, messages from a beam factor to all variables may be computed at one pass. This, again, is possible due to dynamic programming.

### 2.4. Approximate front-based solution

As stated above, the iteration over all front shapes takes \(O(D^T)\) steps. This may still be too expensive for wide beams. We further notice then that most of the shapes are highly improbable, and the corresponding fronts do not contribute to (most) factor messages. We thus limit ourselves with exploring only the front shapes near some “most plausible” variants. Technically, an assignment \(\bar{a}\) to the front shape \(t^*(d)\) is equivalent to an integer number with \(D\) digits in base \(T\). Given some starting number, we may thus simply consider a few values in its vicinity such that the number of steps is determined by some “search depth” parameter. However, as discussed above, numbers base \(T\) are inconvenient for dynamic programming: on each step, many entries in \(\bar{a}\) may change by more than a single
unit. We thus adopt a parameterisation of the $D$-dimensional search volume with a space-filling Hilbert curve [7]. By following this curve, we are guaranteed that each step changes a single entry by plus or minus one.

The starting front shapes ("seeds") can be obtained from additional sensors or with any heuristic method such as that of [8]. Finally, an efficient message calculation scheme for a beam factor is as follows. First, we select some starting front assignment ("seed") $\tilde{a}$ and compute the value

$$X(\tilde{a}) = E_b(\tilde{a}) + \sum_{d=1}^{D} m(a_d, d) + \sum_{d=1}^{D} \sum_{t=d+1}^{T} \min\left(0, m(t, d)\right).$$

We also initialize all messages to plus infinity. The following assignments $\tilde{a}$ are selected according to the Hilbert curve parameterization. Therefore, $X(\tilde{a})$ and the messages can be updated at each step in constant time as

$$M_{\tilde{a}}(x(t, d)) = \begin{cases} X(\tilde{a}), & \text{if } a_d > t \\ \infty, & \text{if } a_d = t \\ X(\tilde{a}) - \min(0, m(t, d)), & \text{if } a_d < t \\ \infty, & \text{if } a_d > t \\ X(\tilde{a}) - \min(0, m(t, d)), & \text{if } a_d < t \end{cases}$$

For an efficient implementation, caching of minimum values for each cell $(t, d)$ is advised. The complexity of this method depends only on the "search depth" near the seed value and must be determined based on the desired inference accuracy. The remaining infrastructure of the reconstruction framework can be directly inherited from the RayMRF model. We expect that the resulting algorithm will for the first time deliver accurate Bayesian shape estimations based on sonar data.

3. CONCLUSION

In this contribution, we suggest a novel method to reconstruct seafloor shape from side-scan sonar data that is based on the RayMRF model borrowed from the domain of computer vision. For the introduced beam factors, we discuss the core inference step and demonstrate a practically feasible approximate solution that uses dynamic programming and space-filling curves to drastically reduce the message update complexity. In the future, we plan to present the applications of the method to synthetic and real sonar data and quantify its accuracy.

The suggested method does not rely on heuristic treatment of the side-scan sonar data but honors the physical origins of the signal. Employing that method, AUVs equipped only with classical side-scan sonars might reconstruct the bottom surface. Additionally, AUVs equipped with sensors that produce distance measurements (like multi-beam echo-sounders (MBES) or interferometric sonars) can use those measurements as prior knowledge and refine their output with the (typically higher-resolution) data from the imaging side-scan sonar. As a means for a more exact environment mapping, the method should also facilitate the application of simultaneous localization and mapping (SLAM) methods for AUVs to improve the navigation accuracy.
REFERENCES


Posters
DIRECT-SEQUENCE SPREAD SPECTRUM UNDERWATER ACOUSTIC COMMUNICATIONS WITH TURBO EQUALIZATION IN TIME-VARYING CHANNELS

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Abstract: A receiver combines direct-sequence spread spectrum (DSSS) and turbo equalization with MAP algorithm has a potential to provide good performance in underwater acoustic communications. Channel estimation is an essential part in branch metric computation for the state transition in MAP. In time-varying channels, multi-path changes rapidly. Conventional MAP algorithm does not update the channel estimation during information detection process, hence the performance degrades a lot. In this paper, data-dependent update of channel estimation is developed, which is derived from per-survivor processing (PSP). PSP establishes a trellis, traverses all the state and picks survivor paths, which can be used for channel estimation. Because no training data is needed, it can update channel estimation anytime during the information detection process, that is a great advantage in time-varying channels. PSP establishes the same trellis as MAP does, the method can be applied to the receiver in this paper. By comparing current branch metrics of all states, the variation of channel is measured, then time to update channel estimation is decided. Simulation result shows that the receiver using data-dependent update of channel estimation is a good solution for underwater acoustic communication system in time-varying channels.

Keywords: time-varying channel, data-dependent update of channel estimation, turbo equalization
1. INTRODUCTION

Turbo Equalization with maximum a posteriori probability (MAP) algorithm provides excellent performance in wireless communications\textsuperscript{1}. Due to long multi-path spread, the computation complexity is extremely high in underwater acoustic (UWA) channel when regular modulation is applied. The combination of turbo equalization and direct-sequence spread spectrum (DSSS)\textsuperscript{2} reduces the number of symbol contaminated by inter-symbol interference (ISI), hence the computation complexity. Also better performance can be achieved than either turbo equalization or DSSS is used separately.

Channel estimation is an essential part for branch metric computation in MAP. Conventional turbo equalization uses training data for channel estimation, and updates the estimation in certain interval, hence performance degrades a lot in time-varying channel. In this paper, data-dependent update of channel estimation derived from per-survivor processing (PSP)\textsuperscript{3,4} is developed. By comparing the current branch metrics of all states in the trellis, the variation of channel is measured, hence the time to update channel estimation is decided.

As PSP establishes the same trellis as MAP does, this method is also suitable for the receiver combined turbo equalization and DSSS. Simulation is taken to verify its performance, as the result shows, the receiver outperforms conventional turbo equalization in time-varying channels.

The rest of this paper is organized as follow. Section 2 introduces the receiver combined turbo equalization and DSSS, section 3 shows the simulation results, and a conclusion is drawn in section 4.

2. SYSTEM DESCRIPTION

The UWA communication system combines turbo equalization and DSSS is shown in Fig. 1. The transmitted signal is written as

\[ x(t) \]
\( x(t) = \left[ \sum_{n} d_{n} \sum_{l=0}^{L_{c}-1} e^{j2\pi f_{l} t} \right] e^{j2\pi f_{c} t} = \sum_{n} d_{n} f(t - nT_{b}) e^{j2\pi f_{c} t} \)  

(1)

where \( \{d_{n}\} \) denote information sequence after coding and interleaving, \( \{c_{l}\} \quad l = 1, \ldots, L_{c} \) is the spreading sequence, \( T_{s} \) and \( T_{c} \) are the symbol duration and chip duration respectively, \( q(t) \) represents the pulse shaper and \( f_{c} \) is the carrier frequency. The spectrum spreading symbol is denoted as \( f(t) \) for short.

Assume the UWA channel is a time-varying channel with \( N_{p} \) paths

\[ h(t, \tau) = \sum_{p=1}^{N_{p}} A_{p} \delta(\tau - (\tau_{p} - a_{p} t)) \]  

(2)

the amplitude \( A_{p} \) is a constant, the delay equals to \( \tau_{p}(t) = \tau_{p} - a_{p} t \), where the Doppler scaling factor \( a_{p} \) varies from path to path, it makes the channel time-varying. With \( w(t) \) denote the additive noise, received signal in base band can be written as

\[ y(t) = \sum_{p=1}^{N_{p}} A_{p} e^{j2\pi f_{l} t} \sum_{n} d_{n} f((1 + a_{p} t - \tau_{p} - nT_{b}) e^{j2\pi f_{c} t} + w(t) \]  

(3)

The main procedure of the receiver is iterative equalization and decoding which is the same with conventional turbo equalization. As shown in Fig. 1, a PSP based branch metric calculation module is added before SISO equalization. As trellis based algorithm, PSP and MAP establish the same trellis, hence the branch metrics calculated by PSP can also be used in the equalization. Data-dependent update of channel estimation is applied to decide the time for channel estimation, hence more accurate branch metrics can be achieved than conventional turbo equalization where channel estimation is updated in certain time interval. Hence, the performance improves in time-varying channel.

![Fig. 2 Trellis used in proposed receiver](image)

Fig. 2 shows the trellis of the receiver. Let \( \tau_{\text{max}} \) denote maximum channel delay spread, each symbol induces ISI to the next \( L = \lceil \tau_{\text{max}} / T_{b} \rceil \) symbols. The states at time \( k \) is define as \( S_{k} = \{d_{k-L}, \ldots, d_{k}\} \), with the alphabet size of \( M \), total number of state in the trellis is \( I = M^{L} \), as shown in vertical line in Fig. 2. \( \gamma(S_{ni}, S_{m}, k) \) denote the branch metric for state transition between \( S_{ni} \) and \( S_{m} \), which is defined as

\[ \gamma(S_{ni}, S_{m}, k) = \|z_{k} - \hat{z}(S_{ni}, S_{m}, k)\|^{2} \]  

(4)

where \( z_{k} \) is vector of received signal \( z_{k} = \{y_{(k-1)i}, \ldots, y_{ki}\} \), and \( \hat{z}(S_{ni}, S_{m}, k) \) is the estimated received signal \( \hat{z}(S_{ni}, S_{m}, k) = \{\hat{y}_{(k-1)i}, \ldots, \hat{y}_{ki}\} \), which is estimated by channel
parameter and transmitted signal. Among the $M$ state transition ended with state $S_m$ in time $k$, the one with least accumulated metric $\Gamma_i(S_m,k) = \Gamma(S_{ni},k-1) + \gamma(S_{ni},S_m,k)$ is picked as survivor branch, as solid line in Fig. 2. The receiver picks survivor branches for all states in the trellis, then corresponding information sequence for each state can be interpreted from these survivor branches, which are used in channel estimation.

It is a great advantage in time-varying channel that PSP estimates channel parameters without using training data. The channel can be estimated every time new symbol received, but it will be highly computation complexity cost. Hence data-depended update of channel estimation which decides the time for channel estimation using received data is developed in this paper. When survivor branches for all $I$ states in time $k$, the corresponding branch metrics are listed as \( \{ \gamma(S_{ni},S_{i},k), \gamma(S_{ni},S_{2},k), \ldots, \gamma(S_{ni},S_{I},k) \} \). $I$ possible sequences can be interpreted from these survivor branches, only one of them is the same with source information sequence, assume it is the one ended with state $S_m$.

When the channel parameters are estimated correctly, the estimated received signal related to $S_m$ is very close to the actual received signal, hence branch metric $\gamma(S_{mn},S_m,k)$ is close to 0, other $I-1$ branch metrics will be much larger. Otherwise, when the channel parameters are incorrect, no estimated received signal is close to the actual received signal, hence the difference between $\gamma(S_{mn},S_m,k)$ and other branch metrics shortens. So the ratio between $\gamma(S_{mn},S_m,k)$ and average value of other branch metrics is defined as a criterion for accuracy of channel estimation

$$
\delta_k = \frac{\gamma(S_{mn},S_m,k)}{\text{mean}(\gamma(S_{ni},S_i,k)|i=1,\ldots,I;i \neq m)}
$$

When $\delta_k$ is smaller than the threshold, estimated channel parameters are suitable for the calculation; otherwise, channel estimation for all states should be updated, where survivor branches are used to provide training data. In this way, we update the channel estimation when UWA channel changes. During the process, the decision is made based on the received data, the update of channel estimation is all data-depended.

After all branch metrics are calculated, the SISO equalization in Fig. 1 processes as conventional turbo equalization does, only more accurate branch metrics are used.

3. SIMULATION RESULTS

The simulation system works on carrier frequency of 6kHz, the bandwidth $B = 2kHz$, hence the chip interval is $T_c = 0.5ms$, with a spreading factor $L_c = 31$, the symbol interval is $T_b = L_c T_c = 15.5ms$. The maximum delay spread of the channel is $\tau_{max} = 25ms$, so ISI length $L = \lceil \tau_{max} / T_b \rceil = 2$. As binary phase-shift keying (BPSK) is used, the number of states in trellis equals $I = 4$. Channel Code is (13,15) recursive systematic code (RSC), random interleave with a frame length of 1024 is used.

Channel model in formula (2) is used in the simulation. Number of multi-path $N_p$ is 10, amplitude $A_p$ is Gaussian distributed, and delay $\tau_p$ is randomly generated with a uniform distribution on the interval $[0, \tau_{max}]$. Two kinds of channel are used in the simulation, static channel and time-varying channel. In static channel, Doppler scaling factor $a_p$ are all set to 0, while in time-varying channel, $a_p$ is randomly generated from a
uniform distribution on the interval \([- v_{\text{max}} / c, v_{\text{max}} / c]\), where \(v_{\text{max}}\) is the maximum Doppler speed set to 4m/s and \(c\) is the sound speed in water.

The effectiveness of data-depended update of channel estimation is verified. Fig. 3 shows an example of how \(\delta_k\) changes with time in the two channels. Threshold for update of channel estimation is set to 0.5. As we can see from the left figure, \(\delta_k\) fluctuates in static channel, but stays in small value and does not exceed the threshold. In comparison, \(\delta_k\) goes up rapidly in time-varying channel as showed in the right figure. It takes round 11 symbols to exceed 0.5, and goes down to round 0.1 immediately after the channel parameters are updated. This example shows that data-depended update of channel estimation is capable to detect the change of channel and update the estimation.

Performance of the proposed receiver is also examined, Fig. 4 shows the simulation results, performance of conventional turbo equalization is also provided as a comparison.

Simulation result in static channel shows in the left figure. As the channel does not change with time, data-depended update of channel estimation does not have much advantage against conventional turbo equalization. The two receivers have almost the same performance. In time-varying channel, because conventional turbo equalization does not update channel parameter during the process, calculation of branch metrics is not accurate, hence performance degrades a lot, with an error floor on \(3 \times 10^{-3}\), as dotted line showed in the right figure. In comparison, as data-depended update of channel estimation is used, the proposed receiver updates channel parameters when the channel changes, more accurate branch metrics are provided, its performance degrade little. Only around
1dB difference in SNR is observed on the level of $10^{-4}$ compared to the performance in static channel.

The simulation result proves that the proposed receiver works well in time-varying channel with difference delays and Doppler scales for each path.

4. CONCLUSION

In this paper, a receiver combines DSSS and turbo equalization is proposed. Conventional turbo equalization does not update channel parameters during the process, its performance in time-varying channel degrades a lot. In order to solve this problem, data-depended update of channel estimation derived from PSP is developed. By comparing all the branch metrics in current time, a criterion marks the accuracy of channel estimation is defined. In time-varying channel, the value of this criterion changes with the channel, receiver updates channel estimation based on this value. Simulation result shows that the proposed receiver works well both in static and time-varying channel, while error floor is observed for conventional turbo equalization in time-varying channel.

Setting criterion threshold is a critical part of the receiver, which has a direct impact on system performance. The setting of threshold depends on several factors, including SNR of received signal, number of states and the multi-path of UWA channel. The impact of these factors on setting threshold is still under investigation, there is no direct formula between them. More results should come as the investigation continued.

REFERENCES

UNDERWATER ACOUSTIC COMMUNICATION SYSTEM SIMULATION BASED ON GAUSSIAN BEAM METHOD

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\textbf{Abstract:} Due to dynamic multipath propagation structure and Doppler spreading, the underwater acoustic channel make the performance of underwater communication systems seriously degraded. Comparing with costly sea trials, underwater acoustic channel models provide a reliable and cheaper tool for predicting the performance of communication systems. The accuracy and efficiency of the channel model is essential for prediction. The ray theory based on Gaussian beam method has merits of high efficiency, clarity in physical meaning, and being easy to be parallel processing. In this paper, we proposed a novel simulator based on Gaussian beam method to estimate impulse response function of underwater mobile communication systems. Simulator operates periodically. At the input port of the simulator, the evaluated communication systems will transmit a frame of communication signal every fixed intervals. According to the relative position relation between transducer and hydrophone in the acoustic field, simulator will estimate the current impulse response function. At the output port, a frame of communication signal will be acquired and processed with a mosaic method. When transmission is over, the BER will be calculated. Parameters such as depth, sound profile, SNR and SL of transducer can be set up in the simulator. Simulations and trial are performed to validate this method.
The results indicate that it help to assess the performance of underwater communication systems different in modulation method.

**Keywords:** underwater acoustic communication; multipath propagation; Doppler spreading; ray model; simulation of underwater acoustic communication.

**INTRODUCTION**

As electromagnetic waves cannot propagate over long distances in seawater, acoustics is the favorite technology for this specific scenario.

Yet, underwater is a challenging environment for communication. The main reason lies: 1. Acoustic wave propagation is strongly inhomogeneous because of irregular sea boundaries, as well as variations in sound speeds over a section of ocean. 2. Doppler spread due to fluctuations in the environment or relative motion between the transmitter and the receiver, sea surface, etc. As a result, the acoustic signals are affected by time varying multipath, which may create severe inter symbol interference (ISI) and large Doppler shifts and spreads. These highly space, time and frequency dependent features pose numerous obstacles for any attempts to establish reliable and long-range underwater acoustic communications.

Recently, the necessity of underwater acoustic communication and demand for transmitting and receiving various data such as voice or high resolution data are increasing as well. UNDERWATER ACOUSTIC (UWA) communication systems have to be designed to operate in a variety of conditions that differ from the nominal ones due to the changes in system geometry and environmental conditions.

As ocean deployment can be expensive and quite difficult, it may not often be practical to perform extensive field testing prior to full deployment. To allocate the appropriate resources (power, bandwidth) before system deployment, as well as to design appropriate signals and processing algorithms, it is necessary to have a relatively accurate channel model.

The contribution of this work is the development of an improved communication model Based On Gaussian Beam Method. This model allows for generation of parameters to be based on a particular target environment, rather than a general set of characteristics. Furthermore, we utilize data from a lake trial to verify the proposed model.

This paper is organized as follows: Section II describes related work and provides motivation for an improved underwater simulation model; Section III describes the proposed model and provides the methodology used in its development; Section IV describes the verification of the model; Section V provides results; and Section VI provides conclusions and indicates areas for future consideration.
I. BACKGROUND

A. Related Work

The proposed model is built upon the BELLHOP beam tracing program developed by Porter [1], as implemented in the Acoustic Toolbox [2]. The BELLHOP program computes the acoustic field by tracing the paths of beams as they leave a source. The central ray of a beam follows the standard equation,

$$\frac{d}{ds} \left( \frac{1}{c(r, z)} \frac{dr}{ds} \right) = -\frac{1}{c^2(r, z)} \nabla c(r, z)$$  \hspace{1cm} (1)

where $r$ is the range and $z$ is the depth using cylindrical coordinates, $s$ is the arc-length and $c(r, z)$ is the speed of sound at a particular point.

The BELLHOP program provides path computations that connect a given source-receiver pair. These paths are dependent on a description of the environment provided by the user. These paths represent direct, reflected, and refracted routes that acoustic waves follow [3] [4].

B. The Need for an Improved Model

Some of the existing simulation models are based on overly simplified conditions that treat the ocean as a whole, the others normally do not consider any unique attributes of a specific area in the ocean. Since underwater channel characteristics are highly space-time variable, it is not easy to build a physical model in which all phenomena are considered.

Beam tracing tools, such as Bellhop, use ray theory to provide an accurate deterministic picture of a UWA channel for a given geometry and signal frequency, but they do not take into account random channel variation.

In order to emulate mobile communication systems, in our model time-indexed channel impulse response (CIR) will be called for to process the communication data packet according to the relative position relation between AUV-receiver pairs.

II. MODEL DEVELOPMENT

A. Introduction to the Proposed Model

As the ocean environment varies greatly with geography [4], it is desirable to tailor the simulation model to the area of target deployment. The environmental parameters factored into the initialization of the model are the Sound Speed Profile (SSP), the contour and roughness of the bottom, the sound propagation characteristics of the sea-floor and surface half space, and the range of positions of any possible source and receiver nodes.

Once parameters have been collected, the model will provide physical layer modeling for a simulation package. The model will provide functions that will return
path loss, propagation delay, and bandwidth for a given transmitter-receiver pair, as well as the calculation of the effective distance of a transmitted signal.

B. Development

The proposed model derives its parameters from an analytical model that can provide an accurate prediction of channel response. In our research, the BELLHOP beam tracing program was used [3]. This model is an efficient ray tracing tool for performing two-dimensional analysis of an ocean environment. The BELLHOP program was selected because of its efficiency and accuracy in generating arrival data consisting of amplitude, phase, and delay for each path between a given transmitter-receiver pair. The proposed model then processes these arrivals to generate the required delay, attenuation, and signal fading characteristics. The BELLHOP program is used as a preprocessing tool to provide data to the next step.

1) Environmental Description: To best utilize the ray tracing capabilities of the BELLHOP program, a precise description of the physical characteristics of the target environment is important: These are the depth of the section of the ocean, information concerning the bottom contour and roughness, and information about the surface.

2) Path Collection: With all the environmental parameters set, the BELLHOP program is then used to collect the multipath information for each possible link. In order to gather data for the entire target area and to determine how the acoustic signals will be received for any transmitter-receiver pair, a series of arrival statistics are collected. The input to the model includes the operational frequencies and the possible locations of all transmitter-receiver pairs. The transmitters are defined by their depth, while the receivers are described with their depth and relative horizontal displacement from the transmitters. The selection of frequency range and positions is completely dependent on the communication system’s specifications. The number of paths and delays will remain constant for a certain SSP and a given transmitter-receiver pair, but the attenuation is frequency dependent.

C. the procedure of Simulation:

First, construct a 2-D coordinate system. According to the velocity of AUV and its track, compute the relative position relation between AUV-receiver pairs via time and save these data in a chart.

Second, repeatedly call for BELLHOP program to compute the channel impulse response (CIR) between AUV-receiver pairs in different location and save these time-indexed function in the chart.

Third, setup up parameters of underwater mobile communication systems. The parameters include modulation mode, carrier frequency, data rate, sampling frequency and so on.

Four, run the simulator after the data file for transmission is chosen. According to modulation mode, data file will be decomposed to segments, be transformed to a wave form and then put into data packets. Synchronization code and training code are also added to every packets. Utilizing the look-up table, simulator will compute the convolution of the input stream and the channel impulse response (CIR).

Five, according to SNR, noise is added to every packets.
Last, with a mosaic method all packets will be integrated into a whole data file for processing by receiver and the BER will be calculated.

Although BELLHOP’s preprocessing is time-consuming, higher resolution can be obtained.

IV. MODEL VERIFICATION

We executed the simulation for estimate the performance of the QPSK technique in underwater mobile communication. Data from a lake trial is utilized to verify the model. The lake trial has been done in QIANDAO LAKE. The right picture in Figure 1 shows the satellite picture of the trial site from google. It can be deduced that underwater circumstance in QIANDAO LAKE is complicated. Thus, it’s difficult to use parameters to describe the bottom’s feature.

The experiment of QPSK technique was performed in underwater channel between an AUV and a 10 element Vertical Receiver Array (VRA). The bottom picture of Figure 2 shows the configuration of lake trial. The velocity of AUV is about 0.5m/s. Carrier frequency is fc = 10 kHz, data rate is 5 kbps and sampling frequency is 100 kHz. Transmitter signal is designed as the top picture of Figure 2. The chirp signal is used for synchronization. We transmit the training symbol before the data symbol to modify the coefficient of line equalizer. Fig. 3 shows the diagram of channel multipath structure at different positions using the method of match filter and the corresponding CIRs in position 1 and position 8. In our model, SSP acquired in the lake trial is used for Bellhop program. The top boundary is described as a vacuum. In order to simplify the model, the bottom boundary is treated as perfectly rigid. The SSP of this target location is shown in the left picture in Fig. 1. Fig. 4 shows the Rays calculated by Bellhop in the trial site. Fig. 5 shows the CIRs in different positions calculated by simulator.

Figure 1 the satellite picture of the trial site and the SSP
Fig. 2 The configuration of lake trial and The formation of data packet

Fig. 3 the diagram of channel multipath propagation structure at at different positions and the corresponding CIRs in position 1 and position 8
Fig. 4: The Rays calculated by Bellhop in the trial site.

Fig. 5: The CIRs in different positions calculated by simulator.

III. V. RESULTS

To gauge the value of the proposed model, comparisons were performed. Fig. 3 shows the CIRs calculated by the data from the lake trial and Fig. 5 show the CIRs calculated by simulator. It’s inferred that the difference in CIRs is caused by bottom’s
feature. The variations of CIRs in different positions are obvious both in Fig.3 and Fig.5 which manifest the highly space and time dependent features of underwater acoustic channel.

IV. VI. CONCLUSIONS AND FUTURE WORK

Comparisons lake trial results with simulator results indicate that the proposed model can do help for testing a mobile communication system deployed in a specific area in the ocean. The proposed model provides a great tool for understanding the nature of communication links in an ocean environment. Future work will include extending the model with a more detailed representation of the time-variability of a particular ocean location.

REFERENCES

THE STUDY OF TIME DELAY ESTIMATION TECHNOLOGY BASED ON THE CROSS-SPECTRUM METHOD

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Abstract: In Many fields of modern radar and sonar, parameters of the target distance and azimuth often need to be measured accurately, one of its key technologies is the time delay estimation, time delay estimation directly affects the accuracy of acoustic positioning effect. Traditional delay estimation consists mainly of the generalized correlation method, phase-spectrum analysis, parametric model estimation, and adaptive time delay estimation. Cross spectrum method is a common method of time delay estimation, the method in high noise environment, can obtain more accurate estimation of delay; but in low SNR environment, the performance of this method in sharp decline. Cross spectral method is first transformed into the frequency domain, in order to get higher accuracy, often need to segment average, one estimation of the time delay often need thousands of points or even thousands of data points, and the individual outliers will make a serious decline in accuracy. In order to reduce individual outliers brought by the nonstationarity of signal, improve the accuracy of estimation, this paper presents an improved cross spectrum time delay estimation method, theoretical analysis, and gives the specific implementation steps. Computer simulation results show that, the improved method in the low SNR environment, can improve the estimation precision. Cross spectrum time delay estimation method is improved, the optimization in the conventional cross spectral method, less computation complexity, and has a strong practical.

Keywords: Passive positioning, time delay estimation, cross-spectrum method
1 INTRODUCTION

In many fields of modern radar and sonar, parameters of the target distance and azimuth often need to be measured accurately, one of its key technologies is the time delay estimation. For passive sonar system, the way to get the target distance and azimuth is using a sensor array. Through the detection of target sound, it calculates the delay of each sensor signal difference, and gets the distance value according to the geometric positioning principle. Time delay estimation is the foundation of passive acoustic localization, its accuracy is directly related to whether can meet the practical requirements of positioning accuracy. It is also the foundation of extraction of information technology such as multiple paths separation, feature extraction, classification and recognition.

Cross-spectrum method is a common method of time delay estimation, the method in high noise environment, can obtain more accurate estimation of delay; but in low SNR environment, the performance of this method in sharp decline. Aiming at this problem, this paper presents an improved cross-spectrum time delay estimation method, theoretical analysis, and gives the specific implementation steps. Computer simulation results show that, the improved method in the low SNR environment, can improve the estimation precision, less computation complexity, and has a strong practical.

2 INTRODUCTION OF CROSS-SPECTRUM METHOD FOR TIME DELAY ESTIMATION

The basic principle of correlated signal processing is the statistical characteristics of the signal and interference (correlation) difference to improve the output SNR of receiving system. If the source to satisfy the far field condition, in the presence of noise conditions, the mathematical model can be established with two sensors spatially independent detection as follows.

\[ x_1(t) = s(t) + n_1(t) \]
\[ x_2(t) = s(t - \tau) + n_2(t) \]

\( x_1(t) \) and \( x_2(t) \) are the received waveforms and two hydrophones, \( \tau \) is the time difference between two hydrophone signal. \( s(t), n_1(t) \) and \( n_2(t) \) are the stationary random process. The assumption that the signal \( s(t) \) and noise \( n_1(t) \) and \( n_2(t) \) are uncorrelated, usually the time delay estimation is to be completed on the estimation of \( \tau \). To determine the time delay and the common method to determine angles of arrival of signals is calculating the cross-correlation function. The following diagram, gives the basic structure of the cross-correlation time delay estimation:
The two signals as an example, its correlation is:

\[
R_{s\bar{s}}(\tau) = E\left[ x_1(t)x_2(t) \right] = E\left[ s(t)s(t - \hat{\tau}) \right] = R_{ss}(\tau - \hat{\tau})
\]

The \( E \) is expectation. The assumption that the observation time is \( T \), the estimation value for the ergodic process of orthogonal cross-correlation is:

\[
R_{s\bar{s}}(\tau) = \frac{1}{T} \int_{0}^{T} x_1(t)x_2(t - \tau)\, dt
\]

In the formula, \( R_{ss}(\cdot) \) is the autocorrelation function of the source signals. According to the characteristics of autocorrelation function:

\[
|R_{ss}(\tau - \hat{\tau})| \leq R_{ss}(0)
\]

That is, when \( \tau = \hat{\tau} \), \( R_{ss}(\cdot) \) or \( R_{s\bar{s}}(\cdot) \) to get maximum value, which \( x_1(t) \) and \( x_2(t) \) have the greatest similarity at this time and, taking \( \hat{\tau} = \tau \):

\[
R_{s\bar{s}}(\tau_0) = \max\{ R_{s\bar{s}}(\tau) \}
\]

As the time delay estimation, this is the basic correlation time delay estimation method.

According to Wiener–Khinchin theorem, stationary random signal power spectrum density and autocorrelation function are the Fourier transform each other. So:

\[
X_{12}(f, \tau_{12}) = \int_{0}^{T} R_{12}(\tau)e^{-j2\pi f\tau} \, d\tau
\]

\[
= \langle x_1(f)x_2(f, \tau_{12}) e^{j\varphi(f)} \rangle
\]

In the formula, \( X_{12}(f, \tau_{12}) \) is the cross-power spectral function of the correlation function of \( x_1(t) \) and \( x_2(t - \tau_{12}) \), \( x_1(f) \) and \( x_2(f, \tau_{12}) \) respectively is the Fourier transform of the corresponding time waveform of \( x_1(t) \) and \( x_2(t - \tau_{12}) \), \( \varphi(f) \) is the function of phase frequency characteristic as the cross-power spectrum. The time delay estimation \( \hat{\tau}_{12} \) is:
\[ \hat{\tau}_{12} = \frac{1}{2\pi} \frac{d\psi(f)}{df} \]

It is time delay estimation \( \hat{\tau}_{12} \) is \( 1/2\pi \) of the slope of the cross-spectral phase frequency function. The solution of the slope is done by the least square method.

The least square estimation of single parameters is discussed below:

If an estimator is a single parameter \( x \), and it has gotten \( m \) linear observations of \( x \), the observation equation is:

\[ z_i = f_i x + n_i \quad i = 1, 2, \ldots, m \]

\( f_i \) is the known factor, and \( n_i \) is the observation noise of the \( i \) observations.

In this single parameter case, the estimation rule for least squares estimation is to obtain the estimator \( \hat{x} \), which can make the square of the error between observation value \( z_i \) and the corresponding \( h_i \hat{x} \) is minimum. That is:

\[ J(\hat{x}) = \sum_{i=1}^{m} (z_i - f_i \hat{x})^2 \]

Thus the calculation formula for the least squares estimation \( \hat{x}_{ls} \) under single parameter was obtained by the method of finding the minimum value.

\[ \hat{x}_{ls} = \frac{\sum_{i=1}^{m} f_i z_i}{\sum_{i=1}^{m} f_i^2} \]

In the formula, \( f_i \) is the line corresponding frequency value in the cross spectral function, \( z_i \) is the line corresponding argument.

Only when the value \( \tau_{12} \) is smaller, the estimation accuracy is higher. When the value \( \tau_{12} \) is greater, the estimation accuracy is reduced. In particular, when the phase measurement is multi-value, large time delay estimation is worried for the blur of the phase estimation multi-value.

In order to solve the blur of the phase estimation multi-value, usually, usually there are two steps. At first, the time domain signal processing gets the coarse delay, after compensation of the coarse delay, phase measurement is performed by cross-spectral method,
that is to solve the estimated blur of multi-value, but also can improve the phase measurement accuracy.

3 INTRODUCTION OF THE IMPROVED CROSS-SPECTRUM METHOD FOR TIME DELAY ESTIMATION

The improved algorithm is for the least squares method of two-step method in measuring the time delay estimation. Iterative time delay estimation algorithm is proposed in this paper, can improve the delay estimation accuracy at low SNR conditions, needs less computing.

Based on the time domain signal processing to solve the coarse delay, and compensation coarse delay, the tow signals become \( Y_1(t) \) and \( Y_2(t) \). Then they are transformed into \( Y_1(\omega) \) and \( Y_2(\omega) \) in frequency domain. A real time delay of signals \( Y_1(t) \) and \( Y_2(t) \) is \( \tau \).

Set: \( Z(\tilde{\tau}) = \sum_{\omega} \text{real} \left( Y_1(\omega) \overline{(Y_2(\omega) e^{-j \omega \tau})} \right) \) When \( \tilde{\tau} = \tau \), \( Z(\tilde{\tau}) \) gets maximum value.

Step 1: After correlation signal processing, the time delay value corresponding to the maximum and the 2nd maximum of two signals \( Y_1(t) \) and \( Y_2(t) \) are \( \tau_1^1 \) (\( \tau_1^1 = 0 \)) and \( \tau_1^2 \).

Step 2: For the precision delay correction, real time delay between two signals will set between \( \tau_1^1 \) and \( \tau_2^1 \).

Step 3: frequency domain compensation delay are as follows:

\[
\begin{align*}
Z_1^1 &= \sum_{\omega} \text{real} \left( Y_1(\omega) \overline{(Y_2(\omega) e^{-j \omega \tau_1^1})} \right) \\
Z_2^1 &= \sum_{\omega} \text{real} \left( Y_1(\omega) \overline{(Y_2(\omega) e^{-j \omega \tau_1^2})} \right) \\
Z_3^1 &= \sum_{\omega} \text{real} \left( Y_1(\omega) \overline{(Y_2(\omega) e^{-j \omega \tau_1^3})} \right)
\end{align*}
\]

Comparing \( Z_1^1, Z_2^1 \text{ and } Z_3^1 \), the delay corresponding to the maximum value and the 2nd maximum value respectively is \( \tau_1^2 \) and \( \tau_2^2 \).

Step 4: repeat step 1. Set up after M iterations, the delay corresponding to the maximum value and the 2nd maximum value respectively is \( \tau_1^m \) and \( \tau_2^m \).
4 ALGORITHM SIMULATION

The method has been done in the computer system simulation. The simulation conditions: signal processing frequency band is 500~2000Hz; signal frequency sampling is 10kHz; time delay difference of signal 1 and signal 2 is 0.013455s; 200 separate statistics under different SNR

After measuring the coarse time delay by time domain cross-correlation, and the time delay compensation for the original signal, the conventional cross-spectral method and the improved cross-spectral method are used for processing. The simulation results as shown below:

![Fig.2: under different signal to noise ratio of cross spectrum method and improved cross spectral method results](image)

As you can see, the time delay estimation accuracy of the improved cross-spectral method has been greatly improved compared with the conventional method. With the SNR reducing, the delay estimation accuracy of the conventional method decreased more seriously, and the improved method decreased not obviously. In low SNR condition, the time delay estimation accuracy has the very big promotion compared with the conventional method.

5 CONCLUSION

Cross-spectrum method is a common method of time delay estimation. In high noise environment, the method can obtain more accurate estimation of delay; but in low SNR environment, the performance of this method in sharp decline. Aiming at this problem, this paper presents an improved cross-spectrum time delay estimation method, makes theoretical analysis, and gives the specific implementation steps. Computer simulation results show that in the low SNR environment the improved method can improve the estimation precision, is less computation complexity, and has a strong practical.
REFERENCES


THE STUDY OF PASSIVE RANGING TECHNOLOGY BASED ON
THREE ELEMENTS VECTOR ARRAY

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\textbf{ABSTRACT}: Traditional passive ranging technology based on three elements array is well known. Changes in the curvature of a spherical wave front leads to relative time delay of each primitive. By measuring the relative time delay of each base element, the target range and azimuth are estimated. Delay estimation accuracy, target distance, orientation, aperture of array, array installation accuracy and other factors have an impact on the ranging accuracy. The most critical factor is the delay estimation accuracy. Published research literature on ternary array of passive positioning, mainly focused on the sound pressure information. It is well known that sound wave has both scalar quantity and vector field, while traditional acoustic pressure sensor system merely makes use of its acoustic pressure information. Vector hydrophone, also called combined sensor, is combined by traditional and omni-directional pressure hydrophone and natural dipole independent on frequency, which can co-locating and simultaneously measures pressure(scalar field) and particle velocity(vector field) of acoustic field. This paper presents the passive ranging technology based on three elements vector array, representing Traditional passive ranging technology based on three elements array, more fully applied sound field information to improve the accuracy of delay estimation, and ultimately improve the passive ranging accuracy.

\textbf{Key Words}: Three elements array, Passive Location, Vector sensor
1. INTRODUCTION

There are three main types of currently developed passive ranging sonar, as follows: passive ranging technology based on three elements array, target motion analysis (TMA) and matched field processing (MFP). Passive ranging technology based on three elements array is the object of this thesis, which mainly use the change of wavefront curvature based on spherical wave or cylindrical wave to complete the measurement. By measuring the relative delay of each element, the target distance and direction are estimated.

Delay estimation accuracy, target distance, orientation, aperture arrays, arrays installation accuracy and other factors make an appreciable difference to ranging accuracy. Delay measurement accuracy is the most critical factor. With decreasing distance, changing the curvature of wave front is more and more big, the measurement accuracy of time delay, that is the ranging precision, is also more and more high. This is easy to achieve the high-precision tracking of the rapid short-range goal. It has important implications for improving melee combat capability.

It is well known that sound wave has both scalar quantity and vector field, while traditional acoustic pressure sensor system merely makes use of its acoustic pressure information. Vector hydrophone, also called combined sensor, is combined by traditional and omni-directional pressure hydrophone and natural dipole independent on frequency, which can co-locating and simultaneously measures pressure (scalar field) and particle velocity (vector field) of acoustic field. No doubt, the more the information, the better the signal processing effect will be.

Passive ranging technology based on three elements vector array, compared to passive ranging technology based on three elements scalar array, more fully applied sound field information to improve the accuracy of delay estimation, and ultimately improve the passive ranging accuracy.

2. THE PRINCIPLE OF PASSIVE RANGING TECHNOLOGY BASED ON THREE ELEMENTS SCALAR ARRAY[1]

Assume that the target is a point source, waves rippled outward by spherical wave propagation. The model of passive ranging technology based on three elements array as shown Fig.1.
In Figure 1, S is defined as the sound source. $H_1$, $H_2$ and $H_3$ denote the 3 elements of passive sonar array. Inter-element spacing: $\overline{H_1H_2} = \overline{H_2H_3} = d$. The angle between the target and the y axis is $\theta$. The distance of target to each array element: $\overline{SH_1} = r_1$, $\overline{SH_2} = r_2 = r$, $\overline{SH_3} = r_3$. Among them, $r_2$ is to measure the target distance ($r$). In general, $r$ is greater than $d$. Therefore, only a slight difference between $r_1$, $r_2$, and $r_3$ by the actual calculation results. The tiny difference is that we measure objects.

Just calculate the delay difference ($\tau_{12}$) of $t_1$ (the time required for signal transmission to $H_1$) and $t_2$ (the time required for signal transmission to $H_2$), and the delay difference ($\tau_{23}$) of $t_2$ (the time required for signal transmission to $H_2$) and $t_3$ (the time required for signal transmission to $H_3$), $r - r_1$ and $r_3 - r$ can be calculated. From a purely geometrical point of view, that of $d$, $r - r_1$ and $r_3 - r$, $r$ can be solved out.

Take $H_2$ as the origin of coordinates to establish the coordinate system is shown in Figure 1. Distance from the target to the array elements:

$$
\begin{align*}
  r_1 &= \sqrt{r^2 + d^2 + 2rd \sin \theta} \\
  r_2 &= r \\
  r_3 &= \sqrt{r^2 + d^2 - 2rd \sin \theta}
\end{align*}
$$

(1)

Assuming the speed of sound is $c$, delay difference is respectively expressed, as follows:
\[
\tau_{12} = \tau_1 - \tau_2 = \frac{1}{c}(r_1 - r_2) \\
\tau_{23} = \tau_2 - \tau_3 = \frac{1}{c}(r_2 - r_3) \\
\tau_{13} = \tau_{12} + \tau_{23}
\]  

(2)

Where \( \tau_{12} \) denotes the delay difference between the array element 1 and the array element 2, \( \tau_{23} \) denotes the delay difference between the array element 2 and the array element 3.

\[
c\tau_{12} = \sqrt{r^2 + d^2 + 2rd \sin \theta} - r \\
c\tau_{23} = r - \sqrt{r^2 + d^2 - 2rd \sin \theta}
\]

(3)

(4)

Equation (3) and (4) respectively respectively were removed, and then squaring on both sides of the equation.

\[
2r[c\tau_{12} - d \sin \theta] = d^2 - c^2\tau_{12}^2 \\
2r[d \sin \theta - c\tau_{23}] = d^2 - c^2\tau_{23}^2
\]

(5)

(6)

Dividing formula (5) and (6), the target bearing can be estimated accurately.

\[
\theta = \sin^{-1}\left[ \frac{cd\tau_{13} - c^3\tau_{12}\tau_{23}\tau_{13}}{2d^3 - c^3d(\tau_{23}^2 + \tau_{12}^2)} \right]
\]

(7)

Adding formula (5) and (6), the target distance can be estimated accurately.

\[
r = \frac{d^2}{c(\tau_{12} - \tau_{23})} - \frac{c(\tau_{12}^2 + \tau_{23}^2)}{2(\tau_{12} - \tau_{23})}
\]

(8)

3. THE PRINCIPLE OF PASSIVE RANGING TECHNOLOGY BASED ON THREE ELEMENTS VECTOR ARRAY

In this paper, the program uses a two-dimensional vector sensors, which can co-locating and simultaneously measures pressure \( p \) and particle velocity \( \nu_x, \nu_y \) of acoustic field.

Particle velocity has dipole directivity[2-3].

\[
\begin{align*}
    p(t) &= x(t) \\
    \nu_x(t) &= \cos \theta \cdot x(t) \\
    \nu_y(t) &= \sin \theta \cdot x(t)
\end{align*}
\]

(9)
In addition, the velocity component weighted combination can make the directional rotation in two-dimensional space.

\[ v_c(t) = v_x(t) \cos \psi + v_y(t) \sin \psi = x(t) \cos (\theta_c - \psi) \]  \hspace{1cm} (10)

If \( \psi = \theta_c \), \( v_c \) has a maximum value. By changing the values of \( \psi \), can scan the full range of directivity, which is realized in the horizontal plane directivity of electron spin.

In this paper, the sound pressure and particle velocity appropriate combination to improve the signal to noise ratio of array element domain, the combination of the following form:

\[ 0.5 p(t) + v_c(t) = x(t) \left[ 0.5 + \cos (\theta_c - \psi) \right] \]  \hspace{1cm} (11)

Passive Ranging Technology Based on Three elements Vector Array implementation steps:

Step 1: measure delay difference from each other by sound pressure signal of three elements scalar array;

Step 2: using sound pressure signal delay difference from step 1, according to equation (7), measure target position \( \psi_0 \);

Step 3: from step 2, the calculated position \( \psi_0 \), according to the formula (10) and (11), acoustic pressure, velocity signals are combined for each element of three elements vector array, obtain combined signals;

Step 4: from step 3, processing these combined signals, measure delay difference from each other;

Step 5: using delay difference from step 4, according to the formula (7) and (8), locate the target.

Signal processing flow chart is as follows:

![Signal processing flow chart of passive ranging technology based on three elements vector array](image)

Fig.1: Signal processing flow chart of passive ranging technology based on three elements vector array
4. SIMULATION

The method has been simulated by the computer system. Simulation conditions:
Target distance: 1000m, target azimuth: 30°, the array spacing: 20m, the frequency range of signal processing: 100–3000Hz, signal processing integration time: 1s. Under different SNR conditions, 200 independent statistics.
Simulation results are shown below:

Fig. 2: The comparison between azimuth estimation of vector Array and azimuth estimation of scalar array under different SNR

Fig. 3: The comparison between the ranging of vector Array and the ranging of scalar array under different SNR

The simulation results can be seen: The passive ranging technology based on three elements vector array, representing traditional passive ranging technology based on three elements array, has been some improvement in positioning accuracy. The results demonstrate the effectiveness of the algorithm.
5. CONCLUSIONS

This paper presents the passive ranging technology based on three elements vector array, representing traditional passive ranging technology based on three elements array, more fully applied sound field information to improve the accuracy of delay estimation, and ultimately improve the passive ranging accuracy. This paper propose the passive ranging technology based on three elements vector array, completes the theory analysis, and gives the specific implementation steps. Simulation show that in the low SNR conditions, weak target signal path is enhanced by the proposed method. The improved method can improve the positioning accuracy, less computation required, this method has a strong practical prospects.

REFERENCES


ADAPTIVE DESPECKLING METHOD FOR SAS IMAGES IN NSCT DOMAIN

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Abstract: The synthetic aperture sonar (SAS) is an attractive high-resolution underwater acoustic imaging technique and widely used in seafloor imaging. Due to the coherent nature of scattering phenomena of the SAS imaging, a type of multiplicative noise called speckle affects the further processing and understanding to SAS images. To despeckle in SAS images, a nonsubsampled contourlet transform (NSCT) based adaptive despeckling method is presented in this paper. The NSCT is a flexible multiresolution, multidirection, and shift-invariant image decomposition transform that can be used to separate the speckle from SAS images. In NSCT domain, each high frequency subband is adaptively divided into three types of regions: the targets, the seafloor, and the shallow. Different strategies are selected to despeckle in these three types of regions to reduce speckle while preserve detail information of SAS images. The experimental results of SAS images despeckling show that the proposed method has better performance in speckle reduction and targets contour preservation than other two comparing methods.

Keywords: synthetic aperture sonar, speckle noise, denoising, nonsubsampled contourlet transform
1. INTRODUCTION

Synthetic aperture sonar (SAS) is a widely used high-resolution underwater acoustic imaging technique to acquire high quality seafloor images and detect targets on the seafloor. Speckle is random granular multiplicative noise commonly observed in SAS images, by giving variance to the intensity of each pixel and reducing the spatial and radiometric resolution. It origins due to random interference between the coherent returns issued from the numerous scatterers present within the resolution cell of the system [1]. Speckle degrades SAS images quality and adversely affects targets detection and information extraction from SAS images. Therefore, speckle reduction is an important problem in SAS image processing to improve SAS imaging quality.

A great deal of methods has been proposed to suppress speckle and preserve spatial resolution in SAS imaging and in other coherent imaging systems such as Synthetic aperture radar (SAR) and medical ultrasound imaging. Some spatial filters like Lee, Kun, and Frost filter are capably to remove speckle but they cannot accurately preserve detail information in images. The speckle reducing anisotropic diffusion (SRAD) method [2] is able to preserve and enhance edges in images and smoothing homogeneous regions. Recently, several novel despeckling methods based on wavelet transform [3, 4, 5] have been introduced with considerable success. The methods in [5] are presented to shrink all the coefficients in a given subband according to generalized likelihood ratio (GenLik) estimation.

The nonsubsampled contourlet transform (NSCT) [6] is a new development of contourlet transform based on a nonsubsampled pyramid structure and nonsubsampled directional filter banks. The result of NSCT is a flexible multiresolution, multidirection, and shift-invariant image decomposition that has a fast implementation. Compared with traditional contourlet transform, the NSCT construction enables users to design filters with better frequency selective thereby achieving better subband decomposition. The NSCT has been proven efficient in image denoising and image enhancement.

The structure of NSCT consists in a bank of filters that splits the 2-D frequency plane in the subbands. The NSCT can be divided into two steps: the nonsubsampled pyramid decompositions that ensure the multiresolution property and the nonsubsampled Directional Filter Bank (DFB) decompositions that give directionality. In this paper, the

2. NONSUBSAMPED CONTOURLT TRANSFORM

The nonsubsampled contourlet transform is a fully shift-invariant, multiresolution, and multidirection expansion that has a fast implementation. Compared with traditional contourlet transform, the NSCT construction enables users to design filters with better frequency selective thereby achieving better subband decomposition. The NSCT has been proven efficient in image denoising and image enhancement.

The despeckling results of the proposed method realize both speckle reducing and preserving the targets contours simultaneously and powerfully in SAS images.
Laplacian pyramid filter is chosen as "maxflat" mode and the DFB are "dmaxflat7" modes; the decomposition level is 5, with 4 directional subbands in each level.

3. DESPECKLING IN NSCT DOMAIN

3.1. Speckle Mode in NSCT Domain

A SAS image with multiplicative speckle noise is defined as

\[ I = P \cdot S = P + P \cdot (S - 1) = P + V, \]  

where \( I \) is the SAS image, \( P \) is the speckle-free signal in the image, \( S \) is the speckle noise with mean equal to 1, \( V \) is the equivalent zero-mean additive signal-dependent noise.

After the NSCT decomposition, image \( I \) has been decomposed into a low frequency subband \( a_J \) and \( J \) series of high frequency directional subband banks \( b_j \) \((j=1,2,\ldots,J)\) in different resolutions. For the linear and fully shift-invariant property of the NSCT, equation (1) in NSCT domain expresses that noise \( V \) is a zero-mean additive signal-dependent noise in each NSCT high frequency subband.

In NSCT domain, the speckle mode is given by

\[ I^C = P^C + V^C, \]  

where the superscript \( C \) means in NSCT domain.

3.2. Coefficients Classification in NSCT Domain

The image \( I \) can be defined and divided into three types of regions: the targets region, the seafloor region, and the shallow region. For instance, Fig.1 (A) is a SAS image, the ship contours is the defined targets region, the dark shallow of the ship is the shallow region, and others region is the seafloor. For the fully shift-invariant property of the NSCT, the coefficients of SAS image in NSCT high frequency subbands are also classified into three types of regions as the image \( I \) in the spatial domain.

An adaptive thresholding function with a parameter \( k \) and the standard deviation \( \delta_i \) are used to extract the targets region \( S_1 \) from a SAS image \( I \):

\[ S_1 : (I_{S_1} > k \cdot \delta_i). \]  

The targets region \( S_1 \) includes the important target contours of the image \( I \). For SAS images, the value range of \( k \) is about 3~5. Once the targets region \( S_1 \) is calculated, all the coefficients in each NSCT high frequency subband at region \( S_1 \) have been classified into the targets region of this subband.

The shallow region \( S_2 \) is defined as the region of the shallow of the targets in the image \( I \). Therefore, the region \( S_2 \) is composed with notable small values pixels:

\[ S_2 : (I_{S_2} < th). \]  

In each high frequency NSCT subband, all the coefficients at region \( S_2 \) compose the shallow region.

In addition, the other coefficients outside region \( S_2 \) and region \( S_2 \) compose the seafloor region \( S_3 \).

The values of NCST coefficients are dramatic changes in targets regions, slowly varying with less variance in the seafloor regions, and with notable small values in shallow regions. Different methods in NSCT domain are chosen to remove the speckle in these three regions.
3.3. Despeckling Equations in Three Regions

The presented method is armed to remove speckle noise and preserved useful detail information in SAS images, so different despeckling strategies are chosen to remove speckle in region $S_1$, $S_2$ and $S_3$ respectively. From (2), a speckle-free estimation value $\hat{P}^c$ is calculated from the observed value $I^c$ in each NSCT high frequency subband.

a) An adaptive soft-threshold equation is applied to despeckle in $S_1$ regions.

$$\hat{P}^c = \begin{cases} \text{sgn}(I^c) \cdot (|I^c| - T) & |I^c| \geq T \\ 0 & |I^c| < T \end{cases}$$

(5)

$T$ is the soft-threshold of equation (5) that been calculated as in the application of [6].

b) A Minimum mean square error (MMSE) estimation based equation is used to despeckle in $S_2$ regions.

For no obvious target contours in seafloor regions of SAS images, the Gaussian distribution is assumed for the additive signal-dependent noise $V^c$.

Since all the coefficients within a small window are equally distributed and conditionally independent in these regions, $5 \times 5$ windows are used for local spatial adaption. With two hypotheses: $H_0$ “the signal of interest is absent,” and $H_1$ “the signal of interest is present” (in a given coefficient), the MMSE estimation based spatially adaptive despeckling equation in these regions is proposed to suppress the additive noise [5]:

$$\hat{P}^c = P(H_1 | I^c, z) \cdot E(s^c | I^c, H_1)$$

$$= \frac{\mu \eta \xi}{1 + \mu \eta \xi} E(s^c | I^c, H_1)$$

(6)

where $\mu$ is the prior ratio of signal and noise, $\eta$ is the ratio of conditional densities of signal and noise for the coefficient, $\xi$ is the ratio of conditional densities of signal and noise for $z$. Under the zero mean Gaussian noise assumption for $V^c$, the despeckling equation becomes

$$\hat{P}^c = \frac{\mu \eta \xi}{1 + \mu \eta \xi} \cdot I^c.$$ 

(7)

c) While there is no obvious target contours and speckle in shallow regions, the coefficients in $S_3$ regions are preserved unchanged.

$$\hat{P}^c = I^c.$$ 

(8)

4. EXPERIMENTAL RESULTS ON SAS IMAGES DESPECKLING

The proposed method is tested on a SAS image despeckling experiment. The GenLik method [5] and Speckle reducing anisotropic diffusion (SRAD) [2] are also tested on the same image to assess the performance. The SAS image is depicted in Fig.1 (A). The despeckling results of the proposed method, the SRAD method and the GenLik method are depicted in Fig.1 (B), (C) and (D), respectively.
From Fig.1, it can be observed that the proposed method significantly suppress the speckle in the SAS image, while maintaining the ship contours. The little objects and seafloor texture of the image are also well preserved in Fig.1 (B). Among the results of the three methods, the proposed method shows the improved speckle suppression ability and the best performance of contours and detail information preservation.

Fig.2: Despeckling results on SAS image2. (a) Original image. (b) Result of the proposed method. (c) Result of SRAD. (d) Result of GenLik.
The above three methods are tested on another SAS image. Fig.2 (A) depicts the original SAS image and Fig.2 (B)-(D) show the despeckled images processed by applying the proposed method, the SRAD method and the GenLik method, respectively. The despeckled images demonstrate that the proposed method produces a superior performance result of the visual effort. In Fig.2 (b), all the speckle in the ship and in seafloor area has been successful suppressed. The outline of the ship is well preserved. The proposed method can achieve speckle suppression and important contours preservation simultaneously. The visualization improvement of the proposed method is the most favoured for the following image processing.

5. CONCLUSION

This paper presents a NSCT domain adaptive despeckling method for SAS images. An adaptive classification is applied for SAS images to use different despeckle strategies to targets, seafloor and shallow regions respectively in each NSCT high frequency subbands. Experimental results prove that our method is the most effective in the tested methods both in terms of speckle reduction and in terms of targets information preservation.

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INFLUENCE OF SHIP RADIATED NOISE LEVEL DIRECTIVITY ON THE ASSESSMENT OF UNDERWATER NOISE MAPS

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Abstract: Recent directives outline the need to mitigate underwater noise footprint due to shipping, to prevent negative consequences to marine life. The need is becoming even more acute because of shipping traffic, which increases steadily. In that context, the goal of the EU project AQUO is to provide policy makers with practical guidelines, in order to mitigate underwater noise footprint due to shipping. The retained definition of underwater noise footprint involves three different quantities: (i) the radiated noise level of a ship, which is a physical, measurable quantity, used to characterize a ship as a source of underwater noise; (ii) the underwater noise map due to shipping, which is a physical quantity used to determine underwater noise due to shipping level in a maritime area and (iii) the underwater noise impact on marine life. In the objective of the assessment of the underwater noise footprint, the ship underwater radiated noise patterns must be accurately evaluated. Whether or not accounting for the directivity of underwater radiated noise is a matter of concern, since some ship types present significant directivity in the horizontal plane, even at low frequencies. We consider in this paper three ship types (fishing vessels, merchant ships, fishing research vessels), that show very different directional behaviour. Simulations on various test cases, mixing different ship types and operating conditions, allow bringing into focus situations where horizontal directivity may play an important role in the assessment of the underwater noise maps and on related noise footprint indicators.

Keywords: radiated noise, directivity, noise maps
1. INTRODUCTION

AQUO is a collaborative research project supported by the 7\textsuperscript{th} Framework Programme through Grant Agreement No. 314227, addressing the call FP7-SST-2012-RTD-1 “Transport”. The final goal of the AQUO project is to provide policy makers practical guidelines to mitigate underwater noise footprint due to shipping, in order to prevent adverse consequences to marine life. We remind that the different noise mitigation solutions considered in AQUO Project can be split into two categories: (i) reduction of the noise source itself (\textit{i.e.} the ships) through design requirements or recommendations for new vessels, and if possible for the improvement of existing vessels, and (ii) solutions in relationship to ship traffic control. One of these is the speed reduction of the noisiest ships in the maritime area under study, provided it is associated with a reduction of sound source level of these ships.

The assessment of shipping noise footprint is carried out by a methodology defined in [1], involving three levels of information, shown on Fig. 1: (i) URN (underwater radiated noise) from the noise sources (here the ships sailing in the maritime area of interest); (ii) noise maps and (iii) indicators of the impact of underwater noise on marine life.

![Fig. 1: Relationship between noise footprint, noise sources and noise map](image)

In the objective of the assessment of the underwater noise footprint, the ship underwater radiated noise patterns must hence be accurately evaluated. Whether or not accounting for the directivity of underwater radiated noise is a matter of concern, since some ship types present significant directivity in the horizontal plane, even at low frequencies. We consider in this paper three ship types (fishing vessels, merchant ships, fishing research vessels), that show different directional behaviour. The parametric model allowing for the calculation of URN for these ships, as well as horizontal directivity data is presented in section 2. In this work, simplified (spherical law) propagation is considered. In section 3 simulations on various scenarios are carried out and conclusions on the influence on horizontal directivity on noise footprint indicators are given in section 4.

2. UNDERWATER RADIATED NOISE MODEL, SHIPS HORIZONTAL DIRECTIVITY AND NOISE FOOTPRINT INDICATOR
Underwater noise radiation from a ship is due to numerous sources distributed along the hull, or generated by the propellers. At sufficiently large distances, it will be assumed that this radiated noise can be represented by the noise field radiated by an equivalent single point source, characterized by: (i) a source level noise spectrum $SL(f)$, determined in far-field, back-propagated to a reference distance (1 meter); (ii) a directivity pattern which represents the variation of $SL$ with bearing and/or elevation. Standards are being developed by ISO Committees for the measurement of URN from surface ships in deep waters (see [2] and [3]). Some models have been developed in the past; this includes the ANATRA [4] and RANDI [5] models, or models developed by Urick [6], Ross [7] or Wales & Heitmeyer [8]. Comparison between the characteristics of the different models shows that none of these models can distinguish different categories of vessel, as well as handling dependence of noise with speed, tonnage or size. It is hence decided to develop a specific parametric URN model.

The three main contributors of underwater radiated noise are: (i) internal machinery and auxiliaries, including the propulsion plant; (ii) propellers and (iii) hydrodynamic flow noise. A general representation of URN patterns for a given ship category is defined by equation (1):

$$
SL_{TOT}(f,V,T) = 10\log\left(10^{\frac{SL_{mach}(f,V,L_{ref})}{10}} + 10^{\frac{SL_{prop}(f,V,L_{ref})}{10}} + 10^{\frac{SL_{cav}(f,V,L_{ref})}{10}}\right) + 10\log\left(\frac{T}{T_{ref}}\right) \tag{1}
$$

The two main assumptions made are (i) noise varies with ship size as 10 times the logarithm of the tonnage, which is close to Urick’s model, and also ensures some consistency with dependence of radiated noise with ship length from Randi’s model and (ii) the total URN from the ship, at a given speed, is the quadratic sum (in energy) of three URN components: machinery, propeller, and cavitation. These three different components are next parameterized against ship speed and frequency for each category of ships. The full detail of the technical background and data used for the development of these parametric models can be found in AQUO deliverable D2.1 [9]. As an example, the parametric URN model retained for commercial vessels is given below (see equation (2)).

\begin{align*}
SL_{mach}(f,V,L_{ref}) &= 131 + 15\log V \quad \text{for } f < 150 \text{ Hz} \\
SL_{mach}(f,V,L_{ref}) &= 178 - 22\log f + 15\log V \quad \text{for } f > 150 \text{ Hz} \\
SL_{prop}(f,V,L_{ref}) &= 107 - 5\log f + 50\log V \quad \text{for } f < 80 \text{ Hz} \\
SL_{prop}(f,V,L_{ref}) &= 154 - 30\log f + 50\log V \quad \text{for } f > 80 \text{ Hz} \\
SL_{cav}(f,V,L_{ref}) &= 73 + 10\log f + 60\log V \quad \text{for } f < 50 \text{ Hz and } V > 10 \text{ kts} \\
SL_{cav}(f,V,L_{ref}) &= 124 - 20\log f + 60\log V \quad \text{for } f > 50 \text{ Hz and } V > 10 \text{ kts} 
\end{align*} \tag{2}

Horizontal directivities for the three types of ships considered in this study are taken from a paper of Gaggero et al. [10]. This paper details the methodology used to measure ship horizontal directivity; the standard for the measurement procedure of the ship radiated noise issued by the ANSI [11] has been adopted. The measurement method consists in several passages with the buoy at the side. Actual transmission losses have been calculated.
with a simulation model based on a normal modes method (ORCA [12]) and used to
derive the source levels. As an example, the horizontal directivities in the third-octave
band 125 Hz and for the three ship types are shown in the figure below.

**Fig. 2: directivity data for the three ships considered**

Spherical propagation is assumed in this study for the calculation of noise levels at a
given range from the source. In January 2010, a report [13] concerning the Descriptor of
Good Environmental Status under the EU’s Marine Strategy Framework Directive
(MSFD) for inputs of energy and noise was released. The main output of the report
concentrated in the definition of three indicators. We chose to use in this study the third
indicator from this report, which states that the ambient noise level measured by a
statistical representative sets of observation stations in Regional Seas where noise within
the 1/3 octave bands 63 and 125 Hz (centre frequency) should not exceed 100 dB (re
1µPa).

### 3. INFLUENCE OF HORIZONTAL DIRECTIVITY ON NOISE FOOTPRINT INDICATORS

In order to assess the influence of horizontal directivity on noise footprint indicators we
propose to use a fictitious scenario, in which noise footprint indicators are calculated with
and without accounting for horizontal directivity. The square area is 108x108 nmi²
(200x200 km²); a bidirectional shipping route extends over the x-axis. Ships’ type,
tonnage, position, speed and heading are randomly chosen within acceptable bounds. For
the sake of brevity the full detail of this randomly generated scenario is not given here. As
an example, Fig. 3 below shows the underwater noise map for the third-octave band 63 Hz
for one randomly generated scenario, with and without accounting for horizontal
directivity. The Monte-Carlo method is used to obtain statistics on maritime area where
levels exceeds the limit of 100 dB re. 1 µPa on frequency bands 63 Hz or/and 125 Hz.

Fig. 4 shows the evaluated probability density functions regarding the area where levels
exceed the given limit, for frequency bands 63 Hz, 125 Hz, and both. Mean and standard
development values are reported in Tab. 1. As one can see, in both frequency bands,
accounting for ships’ directivity lower the area where noise levels exceed the limit. It can
also be noted that the frequency band 63 Hz exhibit larger noise levels than frequency
band 125 Hz, and hence dominates the global noise footprint indicator (noise level lower
than 100 dB in both frequency bands). For frequency band 125 Hz, the area where noise levels exceeds the limit is 1140.4 nmi² (point source) and 297.2 nmi² (directional source). The influence of directivity is hence much more important for this frequency band, and depending of the scenario chosen (number of ships, tonnage, and speed) it may play an important role in the evaluation of maritime areas to be protected.

Fig. 3: noise map example; blue: fishing ships, red: research ships, green: merchant ships; black line: isolevel at 100 dB re. 1 μPa²

Fig. 4: evaluated probability density function regarding the area where levels exceed the given limit, for frequency bands 63 Hz (left subplot), 125 Hz (middle subplot), and both frequency bands (right subplot)

Tab. 1: mean and standard deviation values obtained from the statistical analysis

<table>
<thead>
<tr>
<th></th>
<th>Area where levels exceed 100 dB [re. 1 μPa²], frequency band 63 Hz [nmi²]</th>
<th>Area where levels exceed 100 dB [re. 1 μPa²], frequency band 125 Hz [nmi²]</th>
<th>Area where levels exceed 100 dB [re. 1 μPa²], frequency band 63 and 125 Hz [nmi²]</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Mean</td>
<td>Dispersion coefficient</td>
<td>Mean</td>
</tr>
<tr>
<td>Omnidirectional sources</td>
<td>3892.2</td>
<td>22.1 %</td>
<td>1140.4</td>
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<tr>
<td>Directional sources</td>
<td>3309.4</td>
<td>20.8 %</td>
<td>297.2</td>
</tr>
</tbody>
</table>
4. CONCLUSIONS

This paper is dedicated to the assessment of the influence of the horizontal ships directivity on the evaluation of noise footprint indicators. Parametric underwater radiated noise models are derived to calculate the source level for three ships categories, and measured horizontal directivities are used. Spherical propagation is applied to obtain noise levels on the maritime area under interest.

The Monte-Carlo method is used with fictitious scenarios parameters in order to obtain a statistical insight on the influence of horizontal directivity. It is shown that frequency band 63 Hz mainly dominates the global noise footprint indicator. With the parameters chosen in this study, which do not cover all pertinent cases, it can be seen that accounting for horizontal directivity lower the area where levels exceeds a given limit. It has been seen the influence of directivity is much more important for frequency band 125 Hz than for frequency band 63 Hz, and depending of the scenario parameters chosen it may play an important role in the evaluation of maritime area to be protected.

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THE COMPARISON OF TWO WAY SOUND PROPAGATION IN HASHIRIMIZU PORT

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Abstract: Reciprocal sound propagation experiment was done in Hashirimizu port to monitor ocean environment of very shallow area. Acoustic monitoring method has advantage for spatial monitoring with few sensors in the ocean over other monitoring methods. In shallow water area, received signal includes many kinds of refracted and reflected signals. These signals confuse accurate travel time from the received signal. This paper shows effective way for demodulation of M-sequence signal of 4 cycle repetition. Using the middle parts of the repetition signal reduce artifacts caused by demodulation because of the discontinuous parts of M-sequence signal. Summation of the repetition signal reduces the environmental noise in the ocean to get good SN ratio. Peak tracing for travel time detection for temperature and current estimation shows good improvement compare with the result of the previous work. A small amplitude peak time series could be confirmed from the demodulated amplitude. They were small but similar in reciprocal propagation compare to the biggest peak time series. This analysis will be an important step for the very shallow water area environmental monitoring with sound.

Keywords: Reciprocal sound propagation, Travel time, Shallow water, M-sequence
1. INTRODUCTION

Acoustic monitoring of the ocean is very suitable method to monitor the ocean. Authors had treated data of long range sound propagations in the Central Pacific Ocean at the year 2000 to reveal ocean phenomena such as temperature, current and tidal effects [1-4]. Even if sound data indicate some ocean phenomena, there is very little way to confirm the phenomena because we cannot see the result directly because of large scale of the ocean. Authors have worked out a reciprocal sound propagation experiment at the Hashirimizu port in front of the Tokyo bay since 2006. The propagation distance is only 120 m and the depth of the area is around 5 m. As the experimental area is very small, we can easily understand the conditions and strange phenomena at the place compare with the deep sea area. In addition, the biggest problem of experiments at the deep sea is power supply. Experiment period and data acquisition interval become restricted by the buttery capacity. At the coastal area, cable power supply enables the long term experiment. Thus, we can control the experimental place like a tank experiment in laboratory while the experimental results include real ocean conditions. As there were few experiments at shallow water area at the Seto Inland Sea [5], Tokyo Bay [6], and Sagami Bay [7] in Japan to monitor current fields and temperature changes, but in most cases, they were also short period because of power supply and difficulties of experimental place usage.

Fortunately, our experiment is enabling to supply the power with cable continuously and monitor the ocean throughout a year by sound propagation. Although there were some mechanical problems, it is still hard to take continuous data, but it is ready to do stationary measurement. From the results in summer of 2008, the propagation path changed drastically because of the steep thermocline at the surface area [8]. But the first arrival signal and biggest peak of the correlation result was used for analysis. Because of the interferences of the surface and bottom reflections, the shape of correlated signal varied according to the tidal level and water temperature [9]. In this paper, authors try to improve the analysis by using the second and the third signals of the repeated signal for accurate travel time estimation.

Fig.1: Map of Hashirimizu port and sound propagation systems locations. Black circles mean the place of the systems and the star shows the position of the ADCP.
2. EXPERIMENT

The experimental place is Hashirimizu port which face to the Tokyo Bay located in the Miura peninsula in Japan. Figure 1 shows the sketch of the Hashirimizu Port and location of the experiment equipment. Two black circles on the figure mean the sound propagation systems. The sound propagation systems locate on the quay across the propagation sea area with a distance of about 120 m. Hereafter, we call the system on the upside in Fig. 1 as sea side system and the lower side one as land side system. The sea bottom covered with very fine silt and sands so it is very smooth. There is a little hollow at the middle between the systems, but it is almost flat except the hollow.

As the sound propagation range is very short, we consider that the water temperature and salinity is uniform in range. It was confirmed by few measurements before this experiment. But surface temperature has much effect from the sun, vertical change of the temperature should be considered. For that reason, six thermometers were set at the three different depths of the both systems, the surface depth, the depth near the transducers, and the bottom depth. These thermometers took temperature of the location every 5 min.

The same system placed on the both side; sea side and land side. In this experiment, transmitter and receiving hydrophone is different instrument. The trigger circuit switches the function of the system from the sending to the receiving and the receiving to the sending. The trigger circuit is based on the 1 pulse per second signal from the global positioning system (GPS). In this experiment, it is important to get accurate sound propagation time between the two systems. GPS receiver provides a pulse so that the both side system can synchronize its sending and receiving timing. As the sound propagation system cannot operate sending and receiving in same time, the both system change its role for each other. As the role changes every 30 s, the reciprocal sound data are generated every minute. The sending signal is 4 times repetition of 7th order M-modulated signal with a carrier frequency of 12.5 kHz. The received signal is recorded for 300 ms from the beginning of the sending time with a sampling frequency of 1 MHz after amplified and filtered with a band-pass filter.

3. SIGNAL PROCESSING

3.1. M-sequence

After the correlation of the M-sequence, amplitude component of the correlation value becomes very sharp pulse like a pulse compression. As 4 repeated M-sequence was used in the experiment, the demodulated signal includes 4 steep peaks appear with the interval of the one cycle M-sequence signal length, \( T_m \). There are some noises before and after the demodulation signals. Although the M-sequence is repetition signal, the sending signal breaks its repetition at the beginning and ending of the signal. When we see some received signals of the experiment, there are so many noises on the required signal even after the band-pass filtered process. Addition of the second and the third repetition signal reduce the effect of such kind of noises. As the true travel time \( T_r \) cannot understand from the raw received signal, let us decide tentative travel time \( T_{r'} \). Then cut the signal 4 blocks as the length of the one cycle M-sequence signal, \( T_m \) from \( T_{r'} \) as shown in Fig.2. The second and the third block are used for cross-correlation. Before the correlation, add the second and
the third for noise reduction and repeat the summation block for 3 times to make continuous signal. After the demodulation by the replica M-sequence signal, pick up the middle block for the analysis of travel time. If the delay of the biggest large peak from the beginning of the block is $T_d$, therefore travel time $T_r$ must be $T_r' + T_d$. This method reduces the artifact caused by the demodulation compare with the previous method [7].

![Fig.2: Bloc diagram of the signal processing of the received signal.](image)

3.2. Peak tracking

The simple peak tracking method is track the biggest peak because as the first arrival waves travelled the direct way, the attenuation of the signal is the smallest compare with the waves travelled the other ways. The later signals may contain the reflecting loss at the sea bottom or sea surface. But in this experiment, under the condition of high temperature gradient and low tide, direct passes suddenly disappeared. In such condition, peak data does not indicate the travel time of the direct pass. Furthermore, even if the direct pass clearly appeared, coefficient of the correlation is not always the biggest compare with the signals from the other passes. If the temperature of the monitored area changed 24 °C to 25 °C, the difference of the travel time between the systems is about 0.11 ms. Therefore, small time window is installed at the peak tracing. Peak value at a moment is assumed to be in ± 0.06 ms of the peak which was detected in the previous time. All data were filtered to remove high frequency component of carrier frequency, and this made wave shapes more smoothly for easy detection of peak tracing.

4. RESULTS AND DISCUSSIONS

Figure 3 shows the reciprocal time series of the demodulation result from 9th to 12th of August. Each correlation signals were normalized with its maximum value. The dots on the figure are the positions where the wave was convex upward and their colors indicate the intensity of the amplitude. The red peaks around 77.5 may be the first arrival time. There are three features from the reciprocal propagation results. The first, the time series
of the first arrival signal of reciprocal propagation is quite different. And the second, the peak pattern of them are also dissimilar. In the deep ocean experiment, the reciprocal propagation paths are considered as the same. In addition, the receiver and the transmitter is the same device. But only in this year, we use different equipment for the receiver and the transmitter. The transmitter and the receiver were arranged in vertical direction with the distance of about 20 cm. That causes the reciprocal path was not completely same. Furthermore, as almost all of the arrival signals include the effect of bottom reflection, boundary angles at the bottom were different in reciprocal propagation. The difference of reciprocal time series must reflect a difference of travel paths. We confirmed that the time series of reciprocal propagation were highly similar in case of using the same device for sending and receiving in other year. The third feature is that there are small peaks before the biggest peak. The amplitudes of the peaks were quite small, but it was clearly confirm that they were continuously confirmed throughout the experimental period. In general, the biggest peak should be the first arrival signal and they were usually direct path without any surface or bottom reflections. But the small peaks appeared before the biggest ones. When the original signal correlated with the M-sequence replica signal, it was considered as a cycle of repetition signal. Therefore it is impossible to decide which part is the first. As the small peaks plotted before the main biggest peaks, the actual arrival time may be one cycle later. If the actual first arrival time is $T_a$, the arrival time of peaks before $T_a$ become $T_a + T_m$. As the direct path should arrive the first, small peaks should be

![Graph](image)

**Fig. 3:** Time series of the demodulated amplitude intensity map of the signal from (a) the sea side system to the land side system, and (b) the land side system to the sea side system. Peaks are indicated with dots and their color indicated the intensity of the amplitude. Once cycle later than they plotted. Interestingly, the time series of these small peaks were relatively similar pattern in reciprocal propagation compared with the time series of the
biggest peaks. Although it still needs to improve the peak tracing, the reciprocal travel time are similar.

5. CONCLUSION

The data analysis method of the reciprocal sound propagation experiment in 2008 was explained in this paper. Repetition of the M-sequence signal gives reduction of the noise and increase SN ratio. It also has advantage in artifacts of demodulation process over the single signal demodulation. The result of proposed method in the data of the experiment, the artifacts before the first peak seems to reduce compare with the previous method. The peak tracing at the very shallow water is very difficult because the time delay of the direct pass and the reflected pass are very short and the interference of the reflected signals change the attenuations of the received signals. However, the peak tracing method with time restriction of the previous peak tracing result shows a little improvement without the restriction result and became smooth and stable. More accurate peak tracing is required for future works for temperature estimation or current detection.

In the first few years of this experiment, the experiments were carried out only summer season, but it became possible to get sound propagation data throughout the year because of instrumental improvement. Although this experiment itself is very fundamental experiment, stocked data and ocean environment comparisons will be the great helps to expand the experiment in deep sea. In addition the system of the shallow water monitoring will be useful for many ports in Japan to monitor the imminent environment of human life area or security purpose.

REFERENCES

A COMBINED GIS-2DFTT MULTI-PARAMETER ANALYSIS OF VERY HIGH RESOLUTION BATHYMETRIC DATA: A CASE STUDY FROM THE VENICE LAGOON

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Abstract: Coastal and transitional environments undergo strong morphological changes due to natural and anthropogenic pressure. In these environments the bathymetric surveys are extremely important for: a) monitoring of the long term environment evolution; b) managing the changes. The recent technological development of the multibeam systems enables them to achieve very high performances also in the extremely shallow waters of coastal and transitional environments.

The morphology of the extremely shallow Venice Lagoon, that surrounds the historical city of Venice, is changing rapidly. This is due to the relative mean sea level rise and at the same time to the strong human induced modifications that started in the 14th century and that are presently still ongoing. In the year 2013 an extensive bathymetric survey was carried out in the Venice Lagoon. During this survey all the channels of the lagoon were mapped with a Kongsberg EM 2040 DC multibeam echo-sounder (MBES) with a grid resolution up to 5 cm. Due to the need to process this large dataset, we developed a semi-automatic method based on a combined GIS-2DFTT multi-parameter analysis and spectral parameterization of the bathymetric data. This analysis allowed us to identify and parameterize the geometrical characteristics of the main morphological features of the channels, like dunes, scours, crests, troughs and sedimentation areas and to extract the channel bottom roughness.

Keywords: 2D fast Fourier transform, Multi-parameter analysis, Venice lagoon, multibeam echosounder
1. INTRODUCTION

Coastal and transitional environments undergo a constant pressure of an anthropogenic nature since they are often highly populated areas or due to the natural interaction between sea and land (river inputs, subsidence, mean sea level rise, etc.). These environments can rapidly change and there is a strong need of constant monitoring and objective quantification of changes in the seafloor morphology. The lagoon of Venice (the largest lagoon in the Mediterranean Sea with a surface of about 500 km$^2$) is a paradigmatic example of extremely shallow environment. It represents the result of the interplay between the natural evolution trends and the human intervention (e.g. [1]). Its actual configuration is the result of the diversion of the main rivers (Brenta, Adige, Piave, etc.) that were flowing into the lagoon by the Republic of Venice starting from the 14th century and the construction of jetties in the 19th century. In the last century the intensive extraction of ground water and the dredging of a major navigation channel for industrial purposes induced a rapid deepening of the central part of the lagoon (e.g. [2]) and caused the sinking of the historical city of Venice by about 25 cm in a century. Moreover the surface of the salt marshes has drastically reduced (e.g. [3]). Presently major modifications at the inlets are ongoing with the construction of mobile barriers to protect the city of Venice from high waters ([4]). These modifications could cause changes in the lagoon hydrodynamics and morphology.

![Fig.1 The Venice lagoon and the bathymetry of the Scanello Channel (DTM resolution of 0.05m and 5 times of vertical exaggeration). Background pictures: data available from the U.S. Geological Survey.](image)

To monitor these changes, in 2013 we carried out an extensive high-resolution (up to 5 cm) MBES survey within the National Flagship Project RITMARE. During this survey that went on from April to December 2013, we explored all channels of the lagoon covering a total area of 50 km$^2$. We acquired a large MBES dataset, including bathymetric, bottom and water column backscatter data. For the quantitative evaluation of seabed changes there is a need for automatic analysis of large bathymetric datasets. Although statistical methods applied to bathymetric data can provide quantitative information on the variability of the seafloor elevation, spectral analysis gives also information on the periodical properties of the seafloor topography in the frequency domain (e.g. [5-6] and references therein). The improvement of
computational power has allowed new analyses of the digital elevation models (DTMs) based on the two dimensional (2D) Fast Fourier Transform (FFT) ([6-8] and references therein). These studies concentrate mainly on linear rhythmic bedforms that can be found in very energetic sandy environments. In this study we extend the 2DFFT analysis to the complex non linear morphology of a tidal channel in the extremely shallow waters of the northern Venice Lagoon (Fig. 1). To this aim, we try to extract all possible information from the 2D FFT analysis by introducing nine parameters describing the spectral properties of the DTM. After georeferencing the results, we could compare within a GIS all spectral parameters with the morphological features of the channel and among each other. Using the combined GIS-2DFFT multi-parameter analysis and spectral parameterization of the bathymetric data we selected the most significant parameters for the quantitative description of the complex channel morphology.

2. THE STUDY AREA

The Scanello channel (Fig. 1) is a natural tidal channel in the northern Venice lagoon. It flows as a branch of a main navigation canal into a salt marsh area. After a short straight path of about 200 m, the channel bends to the North for about 300 m where it separates into two smaller branches flowing into an extremely shallow tidal flat (< 1 m). In the external bend there is a deep scour and in the inner bank a point bar deposition trend is clearly visible. The erosion-deposition pattern is characteristic of the meandering tidal channels. Bedforms are present in correspondence to the scours with a wavelength ranging from a few meters up to ten meters generally increasing with channel depth. The bedforms connect the deepest part of the scour to the point bar slope with an orientation perpendicular to the tidal flow direction. Another deep confluence scour is present at the channel bifurcation. In the two smaller branches the bedforms are mostly three dimensional. Currents in the channel are of the order of 0.1 to 0.6 m/s.

3. MATERIAL AND METHODS

3.1 Data acquisition and processing

The multibeam data were acquired with a Kongsberg EM2040 Dual-Compact MBES during a survey carried out in April 2013. The MBES was pole mounted on a 10 m long boat with 1.5 m draft. The double-head MBES has 800 beams (400 per swath) and a frequency that can range from 200 to 400 kHz. During the survey the frequency was set to 360 kHz and the data were acquired in equidistant mode, ensuring more than 30% of overlap between adjacent survey lines. A Seapath 300 positioning system was used with a Fugro DGPS correction. The sound velocity was measured continuously close to the transducers with a Valeport miniSVS sensor. Moreover, sound velocity profiles were taken regularly with a AML Smart-X sound velocity profiler.

The tide corrections in all the areas were obtained thanks to the hydrodynamic model SHYFEM ([9]) giving the values of water level in 93 locations of the lagoon. The model computes the sea level at each location (station) using the wind data and the sea level data from all the tidal stations in the lagoon and at the inlets as forcing factors or as data to assimilate. The error of the model in the sea level simulation at a certain station is of about 2 cm. All the corrections are referred to the local datum Punta Salute 1897.

In order to process the data and obtain the DTM file of the channel we used the CARIS Hips & Sips accounting for sound velocity variations, tides and basic quality control.
3.2 The combined GIS-2DFTT multi-parameter analysis

The bathymetric data were gridded with a grid-cell size of 0.05 m and the dataset was then subdivided into 10 by 10 m square windows. Adjacent boxes overlapped by 85% of the window size. This overlapping was chosen in order to limit the loss of resolution as a consequence of the windowing procedure. Following a similar procedure to the one described in [6] and [7], for each window we applied a median filter, detrended and tapered the data. The tapering was done through discrete prolate spheroidal sequences (Slepian sequences) to avoid the spectral leakage. We then calculated the 2DFFT of each sliding window to obtain the relative spectrum. The windows at the borders containing no data points were excluded by the calculation, since artificial inputs as zeros or white noise affect the results of the 2DFFT. For each window, we first calculated the maximum value of the spectral density function $S_{max}$. Then we extracted slices of the spectra every 5 degrees. For each slice the following parameters were computed: the power of the spectrum $\gamma_2$ and the spectral strength $w_2$ as defined in [10], the zeroth and the second spectral moment where the $r$-th spectral moment is defined as follows

$$m_r = \int_0^\infty f^r S(f) df,$$

the spectral width $\nu^2$, the spectral skewness $\gamma = m_3/m_2^{3/2}$. Moreover, we computed the spectral constant $Q$, defined as the ratio of central frequency of the spectrum to the resolution, and the mean frequency $f_{mean} = m_1/m_0$.

The results of the analysis were then georeferenced extracting the central coordinate of each sliding window. The matrices of the parameter values with the relative coordinates were then interpolated and mapped in ArcGIS 10.1. In this way we were able: a) to compare the results of the different parameter maps with the morphological features visible in the bathymetry; b) to compare the parameters among each other to evaluate the information they provide.

4. RESULTS AND DISCUSSION

![Fig. 2: Comparison between the bathymetry (DTM resolution 0.05m, vertical exaggeration 5x) and the different spectral parameters computed over 10m x 10m sliding windows over the](image-url)
bathymetry. Their georeferenced and classified values are draped in transparency on the bathymetry for comparison.

In Fig. 2 we show the bathymetry of the Scanello channel and the georeferenced spectral parameters overlaid in transparency over the bathymetry for better comparison. The maximal value of the spectral density function $S_{\text{max}}$ shows the distribution of the main morphological features. Its highest values indicate where the bedforms of predominant size are located. The power of the spectrum $\gamma_2$ reflects the vertical scale of the roughness. The small values of this parameter indicate the occurrence of small scale undulations, while large values indicate the predominance of large-scale bottom surface irregularities in the studied area. The power of spectrum accounts also for the degree of isotropy of the surface. The large values of $\gamma_2$ correspond to an anisotropic seafloor, the small values to an isotropic one. The spectral strength $w_2$ is related to the horizontal scale of the seafloor morphological features and it is proportional to their size.

The **spectral width** indicates the concentration of different size bedforms around a dominant scale bedform. A large value of the **spectral width** indicates that the predominant bedforms are of one scale, a low value that there is a variety of bedform scales.

As it can be seen by the direct comparison in the profiles in Fig. 3, the zeroth-order spectral moment $m_0$, like $w_2$ reflects the dominant seafloor morphological features. It indicates, where they are dislocated forms having dominant size scales in the investigated area. The map of the second-order spectral moment $m_2$ is very similar to the one of $m_0$ (see also the relative profile in Fig. 3) and does not add further information. The same can be said for the **spectral skewness** whose map and profile mirror the values of $m_0$. The map of $Q$ is almost identical to

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**Fig. 3:** On the left, a zoom of the Scanello channel bathymetry (DTM resolution 5 cm, vertical exaggeration 5x); on the right, the values of the bathymetry and the spectral parameters on the same profile plotted with the black line on the left. The profiles show the responses of the different parameters to the bathymetric variation.
the one of the spectral width, while the mean frequency map almost mirrors the spectral width map. Therefore, the five parameters $S_{max}$, $f_2$, $w_2$, $m_0$, and spectral width are enough to describe quantitatively the complex morphology of the tidal channel.

5. CONCLUSIONS

In this study we applied a combined GIS-2DFFT multi-parametrical analysis to the bathymetry of a tidal channel in the northern part of the Venice lagoon. With this combined analysis we explored the meaning and significance of the spectral parameters and of the roughness spectrum parameterization. We set up a semi-automatic procedure that, with the computation of five spectral parameters, can identify the scale of the morphological features and the degree of seafloor isotropy.

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PASSIVE ACOUSTIC DETECTIONS OF ODONTOCETES IN THE IONIAN AND AEGEAN SEAS, GREECE.

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Abstract: The Ionian and Aegean Seas, both part of the eastern Mediterranean Sea, are habitat for several odontocete species including striped (Stenella coeruleoalba), short-beaked common (Delphinus delphis), Risso’s (Grampus griseus), and bottlenose (Tursiops truncatus) dolphins as well as Cuvier’s beaked (Ziphius cavirostris), and sperm (Physeter macrocephalus) whales. Common dolphins and sperm whales in the study area are listed as endangered in the IUCN Red List of Threatened Animals. Currently, very little is known about the seasonal abundance and distribution of cetacean species in the eastern Mediterranean Sea. This information is crucial for the development of effective protection measures for these animals.

In a first attempt to collect baseline data on the occurrence patterns of odontocetes in Greek waters, two passive-acoustic recorders were deployed in 2008 in the Ionian Sea in the vicinity of Station Pylos (36.8N, 21.6E) and in the northern Aegean Sea in the vicinity of Station Athos (40.0N, 24.7E). These recorders operated for 19 and 11 months, respectively. Preliminary results of the data analysis revealed that delphinid species were present at both locations year-round. As expected, sperm whales were predominately detected in the Ionian Sea. Part of this study is the quantification of the soundscape in both locations and the characterization of the cetacean’s habitat. Future work includes species identification of delphinid whistles using the Real-time Odontocete Call Classification Algorithm (ROCCA) to allow an analysis of species-specific occurrence patterns.

Work supported by the Hellenic Centre for Marine Research and the University of the Aegean, Greece.

Keywords: Passive acoustic monitoring, cetaceans, Ionian Sea, Aegean Sea, sperm whale.
Introduction

The Hellenic Seas, located in the eastern Mediterranean Sea, provide habitat for a great variety of cetacean species. The most common odontocetes species in these areas include the bottlenose (Tursiops truncatus), striped (Stenella coeruleoalba), common (Delphinus delphis), and Risso’s (Grampus griseus) dolphins, the harbor porpoise (Phocoena phocoena), Cuvier’s beaked whale (Ziphius cavirostris), and the sperm whale (Physeter macrocephalus).

Common dolphins, harbor porpoises and sperm whales in the Hellenic Seas are listed as endangered in the IUCN Red List of Threatened Animals. The sperm whale population in the Ionian Sea consists of mature males and social groups of females and calves. Their year round presence signifies that this region as an important nursing and breeding ground (Frantzis et al., 1999). The common dolphin population, also listed in Appendices I and II of the Convention on the Conservation of Migratory Species (Bonn Convention, CMS), has dramatically declined in the past 50 years (Bearzi et al., 2003; Cañadas and Hammond, 2008). Common dolphins are rare in the Mediterranean Sea but remain relatively abundant in parts of the Aegean Sea and the eastern Ionian Sea (Bearzi et al., 2003).

Numerous anthropogenic activities threaten the health of the marine ecosystem of the Hellenic Seas. Marine animals rely on sound to sense their surroundings (navigation and prey detection) and to communicate with conspecifics. Therefore the increasing noise pollution caused by commercial and recreational shipping, oil and gas exploration, and naval sonar exercises is of special concern (Weilgart, 2007). The acoustic pollution can negatively impact cetaceans that are highly dependent on sound for vital behaviors.

Odontocetes produce three types of sounds: echolocation clicks, burst pulses, and whistles. Echolocation clicks are broadband transient sounds. The echolocation clicks of certain species exhibit species specific spectral characteristics and inter click intervals (ICI). Burst pulses are series of clicks produced at a very high rate (short ICI). Whistles are tonal calls that can vary in frequency and duration among species. However this variability often is not significant for all the cases and that makes acoustic species identification a challenge. Examples of these three basic call types are presented in Fig. 1.

Fig. 2: Example of a spectrogram of a sound bite (duration: 4.5 s, sampled at 100 kHz) showing various types of delphind vocalizations recorded at Station Pylos in November 2010.
One of the main challenges associated with the conservation of endangered marine mammal species in the Hellenic Seas is the lack of data on their distribution and abundance (Notarbartolo di Sciara and Gordon, 1997). Past research in the Hellenic Seas utilized primarily visual survey data and focused by and large on the Ionian Sea (Frantzis et al., 2003). Traditional visual surveys of cetaceans are expensive and limited by sunlight and favorable weather conditions. Thus, consistent multi-year data sets covering all seasons are not yet available for the Hellenic Seas.

Here we describe a project that uses a passive-acoustic recording/detection system called the Passive Acoustic Listener (PAL) to study the seasonal occurrence of sperm whales and various delphinids in the Hellenic Seas. This project also includes an analysis of the ambient noise levels at each deployment location, their seasonal variations, and major natural and anthropogenic contributors to the underwater soundscape. This study represents the first year-round acoustic monitoring effort of cetaceans in the Hellenic Seas.

**Materials and Methods**

The PAL is low-duty cycle (1.5%) passive acoustic recording system featuring a sampling rate of 100 kHz. The bandwidth of the recordings covers the frequency range of most vocalizations produced by marine mammals in the study area. The PAL has a limited data storage capacity (~2 GB) which allows 2,200 sound bites (raw acoustic data) of 4.5 s duration (a total of 165 min) to be stored during each deployment. Prior to the deployment a maximum number of sound bites (usually between 5 and 7) to be recorded per day (quota) can be programmed. A detailed description of the instrument and its mode of operation are provided in Miksis-Olds et al. (2010).

Two PALs were deployed as part of the POSEIDON II project (http://poseidon.hcmr.gr/), a monitoring, forecasting and information system for the Hellenic Seas operated by the Hellenic Center for Marine Research (HCMR) (Fig.2). The first PAL was deployed at Station Pylos (36.8° N, 21.6° E) in the Ionian Sea at a depth of 500 m (water depth 1,680 m). The instrument was operated for about 19 months (December 2008 - July 2010). The second PAL was deployed at Station Athos (40.0° N, 24.7° E) at 200 m depth (water depth 400 m). The instrument was operated for about 8 months between November 2008 and July 2009. In total, 560 minutes of underwater recordings were collected at both sites.
The acoustic data sets are being analyzed for marine mammal vocalizations using custom analysis software developed by Oregon State University. An experienced analyst (ND) visually and aurally inspects the sound bites to identify bioacoustic signals (Figs. 2 and 3) of interest.

In the initial analysis, of spectrograms the data are screened visually for sperm whale and delphinid vocalizations which can be easily differentiated (Fig. 3). Delphinid echolocation clicks are generally higher in frequency and feature a significantly shorter ICI than sperm whale clicks. Furthermore, sperm whales don’t produce whistles.

![Fig. 2: Locations of the Stations Pylos and Athos in the Hellenic Seas (red triangles).](image)

Fig. 3: Example spectrogram (upper panel) and waveform (lower panel) of (a) a sperm whale detection at Station Pylos and (b) a delphinid detection at Station Athos.
Presence/absence of sperm whales and delphinid vocalizations is logged for every sound bite to evaluate their seasonal occurrence patterns. In this preliminary analysis delphinids were not classified to a species level. Future work will focus on using the Real-time Odontocete Call Classification Algorithm (ROCCA) (Oswald et al., 2007) developed by Co-PI Oswald to evaluate delphinid species composition at the study sites.

Furthermore, we will analyze the recorded data for seasonal patterns in ambient noise levels at Stations Pylos and Athos and will identify major natural and anthropogenic contributors to the underwater soundscape.

**Concluding Remarks**

The Passive Acoustic Listener (PAL) was originally designed to acoustically observe ocean surface processes and especially rainfall. This study demonstrates that the PAL can be used to cost effectively study vocal cetacean species and anthropogenic noise levels for extended periods of time. The collected data sets provide valuable information on the seasonal occurrence patterns of cetaceans in the Hellenic Seas as well as the ambient noise levels they are exposed to. The results of this study will support ongoing and future cetacean conservation efforts.

**Acknowledgements**

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**References**


SOUND PRESSURE FIELD FOCUSED BY OFF-AXIS APLANATIC STRAUBEL ACOUSTIC MIRROR

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Abstract: Underwater acoustic lenses and acoustic mirrors made an effort on reducing sound attenuation, aberration, and having stability for water temperature change. In the past study, we designed an aplanatic Straubel (AS) mirror. The AS mirror is an aplanatic back-surface mirror. The AS mirror could correct spherical and coma aberrations in result of experiment. However, the AS mirror has a problem. The receiver array is located in front of the mirror, which results in the interruption of incident sound waves, and then we designed an off-axis AS mirror and evaluated the convergence properties using numerical calculations. The off-axis mirror is the effective area of the ordinary mirror that is not hidden by the receiver array. The focal length and aperture of the original AS mirror are both 400 mm. Therefore, the aperture of the off-axis AS mirror is 200 mm. In this report, we made an off-axis AS mirror with silicone rubber and brass, and measured the convergence properties at the frequency of 500 kHz for different incident angles in a water tank. We could not measure at larger incidence than 10° due to the restriction of our experimental apparatus. At large incident angles, -3 dB focal areas moved close to the mirrors. This deformation seems to be caused by field curvature. The beam patterns along the spherical imaginary surface were measured at 0°, 5° and 10°. The peak value at 10° was about 2 dB lower than the value at 0°, but the beam widths of the off-axis mirror were almost the same width at all incident angles. From these results it is confirmed that spherical and coma aberrations are corrected. The side-lobe levels at oblique incidence were below -20 dB and lower than the value at normal incidence.

Keywords: acoustic lens, acoustic mirror, imaging sonar, aplanatic, Straubel mirror
1. INTRODUCTION

Underwater acoustic imaging technology is effective for marine resource detection, coastal security, and maintenance of seashore facilities. Underwater acoustic lenses and acoustic mirrors made an effort on reducing sound attenuation, aberration, and having stability for water temperature change\(^1\), \(^2\). In the past study, we designed an aplanatic mirror\(^3\) and aplanatic Straubel (AS) mirror\(^4\). The AS mirror is an aplanatic back-surface mirror. The AS mirror could correct spherical and coma aberrations in the result of experiment\(^5\). However, the AS mirror has a problem. The receiver array is located in front of the mirror, which results in the interruption of incident sound waves, and then we designed an off-axis AS mirror and evaluated the convergence properties by numerical calculations\(^6\).

In this study, we made an off-axis AS mirror with silicone rubber and brass, and measured the convergence properties at different incident angles in a water tank\(^7\).

2. DESIGN OF OFF-AXIS AS MIRROR

2.1. Geometric design

The design method for an off-axis AS mirror is almost the same as that for an ordinary AS mirror. The off-axis mirror is the effective area of the ordinary mirror that is not hidden by the receiver array. The design method for the ordinary mirror has been described in our previous report\(^3\). The off-axis AS mirror can be made by extracting a part with the area from \(y = 0\) to 200 mm of the ordinary AS mirror shown by dashed lines in Fig.1. The gray and black areas are cross-sectional shapes, and regarded as a refractive layer of silicone rubber and a reflective rigid body of brass, respectively. The relative refraction index between water and silicone rubber is 1.5. The focal length and aperture of the original large mirror are both 400 mm. Therefore, the aperture of the off-axis AS mirror is 200 mm. As shown in Fig.1, there are almost no aberrations from incidence angle 0° to 20° in geometrical analysis.

![Fig.1: Cross-sectional shapes and ray trace diagrams of off-axis AS mirror.](image-url)
2.2. **Evaluation using numerical calculation**

In this subsection, aberration and attenuation using a numerical calculation based on the wave theory are considered and the convergence property is evaluated in terms of the beam width, which determines the resolution of the acoustic images. The details of the three-dimensional calculation method are described in ref. [3].

The calculated sound fields of the off-axis aplanatic Straubel mirrors are shown in Fig. 2. The beam patterns at \( x = 0 \) mm and \( z = 400 \) mm are shown in Fig. 3. The sound speed, density, and attenuation coefficient for the lens made of silicone rubber and water used in the calculation are shown in Table I.

![Figure 2](image1.png)

*Fig. 2: Calculated value of sound power distribution of AS mirror of incident angle 0°, 10°, 20° and 30°. The colored areas have over half power of each focal point.*

![Figure 3](image2.png)

*Fig. 3: Calculated value of beam patterns of AS mirror of incident angle 0°, 5°, 10°, 15°, 20°, 25° and 30° normalized by the maximum sound power of incident angle 0°*
Table 1: Parameters used in the calculation.

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<td>Density (kg/m³)</td>
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<tr>
<td>Attenuation coefficient (Np/m)</td>
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3. EXPERIMENT IN WATER TANK

Schematic view of the experimental equipments is shown in Fig. 2. In the experiment, we used a water tank which has 2 meters wide (y-axis), 3 meters long (z-axis), and 0.6 meters depth (x-axis). A transmitter (RESON TC3029) was put in the center of the mirror and was parallel to the z-axis direction. A receiver (RESON TC4035) was put in a movable arm.

We measured sound field distribution around the focus from incidence angle 0° to 10° at 5° step. We input the sinusoidal burst wave whose pulse length was 5 cycles from a function generator (AGILENT 33120A) to the transmitter. The frequency was 500 kHz. The reflected sound pressure was received by a hydrophone, and passed through an amplifier (NF 5307), a band-pass filter (NF 3628), and an A/D converter (ELMEC EC-6904). We defined the mean square at each point as a measured value. The mirror could be rotated in order to change the incident angle, $\theta$.

The measured sound fields of the off-axis AS mirror are shown in Fig. 5. We could not measure at larger incidence than 10° due to the restriction of our experimental apparatus. Gray area and broken line in Fig. 5 show -3 dB areas around focus, and the azimuth angle...
from the z-axis, respectively. At large incident angles, -3 dB areas move close to the mirrors. This deformation seems to be caused by field curvature.

Figure 6 shows the beam patterns along the spherical imaginary array shown in Fig. 5. Each beam pattern is normalized by maximum value at $\theta = 0^\circ$. The peak value at $\theta = 10^\circ$ is about 2 dB lower than the value at $\theta = 0^\circ$, but the beam widths of the off-axis mirror are almost the same width at all incident angles. From these results shown in Figs. 3 and 4, it is confirmed that spherical and coma aberrations are corrected. The side-lobe levels at oblique incidence are below -20 dB and lower than the value at normal incidence.

**Fig. 5:** Experimental results of sound power distributions of off-axis AS mirror under changing incidence angle.

**Fig. 6:** Experimental results of beam patterns of off-axis AS mirror along the imaginary array shown in Fig. 5.
4. CONCLUSION

An incident sound wave coming into an ordinary AS mirror is interrupted by a receiver array. Thus, an off-axis AS mirror was proposed to solve this problem. We evaluated the convergence properties of the off-axis mirror in water tank experiments.

From the results of the experiment, the off-axis AS mirror showed aplanatic convergence properties within the incident angle of 10°, and the beam patterns showed almost the same width within 10° incidence.

In this report, the convergence properties of the off-axis AS mirror were measured at $\theta = 0°, 5°$ and $10°$ incidence. We need measure the properties at wider incident angle than $10°$, and compare with those of an ordinary AS mirror in near future. And we will measure the 3-dimensional convergence properties of the off-axis AS mirror by rotating the mirror in future.

5. ACKNOWLEDGEMENTS

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REFERENCES

PRELIMINARY INVESTIGATION ON THE POTENTIAL OF USING LOW POWER ULTRASOUND TO INDUCE LOW FREQUENCY VIBRATIONS ON AN IMMERSED OBJECT

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Abstract: In this paper, an investigation is undertaken as to whether low power ultrasonic projectors in water can induce vibration on a soft object immersed in water. Experiments were conducted in a water tank using one transducer emitting an amplitude modulated (AM) pressure wave or two converging transducers each one emitting a sinusoidal wave in the 500 kHz range, while their frequencies differed by 500 to 2000 Hz. Short pulses with a rise time around 7 ns were also emitted. The objective of these experiments was to check whether ultrasound can be used to induce vibrations of fish swimbladder in the range of its resonant frequency (500-2000 Hz). In some experiments focal transducers at 5 MHz were also used. In all cases the beams were aiming at a light surface, made of plastic or nylon, where an accelerometer was mounted for picking up vibrations in the frequency range mentioned above. Results for all different setups, using directional and focal transducers are presented and discussed. Prospects for applying the method in order to cause a sound response from fish or simply study their responsive behavior, is also considered.

Keywords: Tank experiments, Bioacoustics, Radiation pressure.
1. INTRODUCTION

The initiative for undertaking the present investigation was given by the need to create sounds in the low frequency range inside small tanks for the purpose of conducting experiments with soniferous fish. In these experiments sound sources need to be placed in the tank in order to conduct conditioning as well as behavioural experiments with small fish. The fish species under investigation are active in the frequency range of 300-1400 Hz. Underwater sound sources in this frequency range are bulky making the use of such sources inside the former tanks, impractical. As an alternative, ultrasonic underwater projectors, because of their small size, might be used in order to create audio sounds. This is possible but still, the space limitations confine the beams’ propagation distance needed for the related phenomena to be established. Another possibility is to try to induce audio frequency vibrations through ultrasound on another object or surface (i.e. tank wall, or another floating or immersed object), which in turn will act as a secondary sound source. This technique can also be used to kind of poke the fish (through dynamic radiation pressure), by aiming the beam(s) at the fish lateral line in order to make it respond and hopefully produce sound or exhibit some other behaviour worth studying. This principle has been successfully used in the past in order to measure the vibrational response of fish swimbladders, [4].

Another, seemingly unrelated, application will be discussed by the end of this paper. We will report on a simple method of detecting whether an individual fish belonging to a species known to have a swimbladder, actually carries a swimbladder. It does happen that some fish do not develop a swimbladder as they should do. The method is based on the reflection characteristics of a short ultrasonic sound pulse and is described at the end of this paper.

2. SERIES OF EXPERIMENTS

Three groups of experiments (A, B and C) are reported:

A) Equipment used: Two ultrasonic immersion projector beam transducers were used in this experiment (V389, Panametrics). These transducers are piezoelectric elements with
central frequencies around 500 kHz and bandwidths of 200 kHz. The beamwidths of the narrow beams depend on the transducer head sizes. The manufacturer of the transducers does not provide calibration charts and documents for these instruments. Their head diameters were 1.75\" inches. The two ultrasonic immersion projector beam transducers were directed towards the water-air interface, in an angle of ~10° between them, in such distance so that the two beams interact exactly at the thin and light plastic surface which formed the bottom of a floating round can (with diameter of 35 cm), where an accelerometer (ENDEVCO 35A, triaxial piezoelectric accelerometer) was mounted at the inside (Fig.1). One transducer was sending CW wave at 500 KHz while the frequency of the second transducer was changing on every trial, spanning a range from: 500.25 to 515 kHz (actual values tried, were: 500.25, 500.5, 501, 502.5, 505, 507.5, 510, 515 kHz). The accelerometer signal was passed through a signal conditioner (PCB 482A16) and was sampled and recorded via a PCI data acquisition card (SPECTRUM High speed 50 MHz, DAC). The spectra of the accelerometer signals for the cases where the projectors’ frequencies differed by 1 and 2.5 kHz, are shown in Fig. 2. It is evident that the dynamic component of the radiation pressure is detected by the accelerometer. Other similar experiments were also conducted using a small thin plastic cap floating on the water surface or a floating plastic ball with an accelerometer (PCB 352810 ICP), mounted at the bottom of the cap and at the top of the ball, respectively. A Panametrics custom made focal transducer with 5 MHz working frequency was placed underwater facing the cap (or ball) bottom at a distance equal to its focal distance (2.54 cm) and was driven via an arbitrary waveform generator (TTi TGA1241), sending an amplitude modulated signal with 5 MHz carrier frequency, 1 kHz modulating frequency and amplitude of 6 V (p-p). The accelerometer signal was amplified through the signal conditioner mentioned above. We obtained similar results with the ones shown in Fig. 2, where there was also a prominent 1 kHz component.

![Spectrum of signal of file: acceler1.dat up to 1.5 kHz](image1)

![Spectrum of signal of file: acceler5.dat up to 5 kHz](image2)

**Fig. 2: Accelerometer signal spectra. Frequencies differed by 1 and 2.5 kHz, respectively.**

B) At this experiment a thin stainless steel bar (of length 45 cm, width of 3 cm and thickness of 0.6 mm), was suspended with almost half of it submerged in water and having an accelerometer (PCB 352810 ICP), mounted close to the bar’s upper end. The water-air interface was at 34.5 cm from the bar submerged end and the focal transducer was placed horizontally at a distance of 2.54 cm (focal distance), facing the bar and at a level of 12 cm from the water surface. The focal transducer was driven with an amplitude modulated signal having 5 MHz carrier frequency, 1 kHz modulating frequency, while the accelerometer signal was boosted 100 times through the signal conditioner and recorded using 44 kHz sampling frequency. The 1 kHz component is also evident here, as seen in
Fig. 3. Besides that, driving the focal transducer with a tone burst similar to that of Fig. 4, caused the rise up of multiple components in the accelerometer signal spectra due to the excitation of the bar’s eigenfrequencies.

![Fig. 3: Accelerometer signal spectrum of thin metal bar excited by a focal transducer.](image)

C) At this experiment we investigated the possibility of using sound in order to detect whether an individual fish carries a swimbladder. At this stage, instead of using actual fish, we tested the ultrasound based method by looking at backscattering signatures from soft and hard round targets. A directional transducer with a small head (V318, Panametrics), having central frequency around 500 kHz and bandwidth of 200 kHz, was used both to send and receive a short pulse. A 100 V tone burst (its shape shown in Fig. 4), was sent from a pulser-device (Panametrics 5058PR) and its reflection from a hard surface or a submerged air balloon, was recorded to a PC through a high speed Data Acquisition Card (DAC). The width of the pulse sent was about 25 ns (although at the pulse shape Fig. 4, the number of samples is shown). The onset time of the pulse was less than 10 ns (~7 ns). Special developed software undertook the whole process (i.e. triggering the send and receive modes and record the waveform). Sampling frequency was 40 MHz, the receiving signal was low pass filtered (with cutoff frequency set at 1 MHz), and an averaging of 4 waveforms was performed. Results for the two cases are shown in Fig. 5, below. Both waveforms, reflected from a hard and soft surface, respectively, look quite different than the emitted pulse due to the transducer response. We notice that in each case although the received pulse has several local extremes we could always single out one extreme that its absolute value was higher than the rest. When the reflection comes from a hard surface the sign of the strongest peak is negative and vice versa.

![Fig. 4: Tone burst shape emitted from the directional transducer](image)
This fact was confirmed in many trials and based on this pattern, we added few lines of code in the recording software to automatically classify whether a target is hard or soft. We claim that this method is simpler than other similar methods [5], and can be used to detect whether an individual fish actually carries a swimbladder. We plan to test this method using in real fish.

3. DISCUSSION - CONCLUDING REMARKS

Three series of experiments were presented in order to check whether it is possible to induce vibrations on a submerged object and potentially use it to ‘agitrate’ fish in laboratory tanks with the purpose of provoking and studying their response (i.e. the production of sound that some species are capable of [2]), which has otherwise been difficult to detect and record. This can be accomplished through the creation of a time dependent (dynamic) component in ultrasound radiation force [1], by using either two interacting continuous-wave ultrasound beams, with slightly different frequencies, or amplitude modulated ultrasound beams. Pulsed ultrasound beams can also be used. Even though we did not use high intensity beams, our results show that this method could have the desirable results when applied to real fish. At this time, an estimate of the exerted radiation force cannot be deduced, because the manufacturer, for the transducers used, does not provide sensitivity data and the particle velocity at the source position is not known. The authors have already been making efforts to calibrate all available transducer equipment in their lab, [3].

A simple technique for detecting fish with or without a swimbladder, by analyzing the backscattering signature of an ultrasound burst tone, was also proven reliable, but it remains to be tested on real fish targets.

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